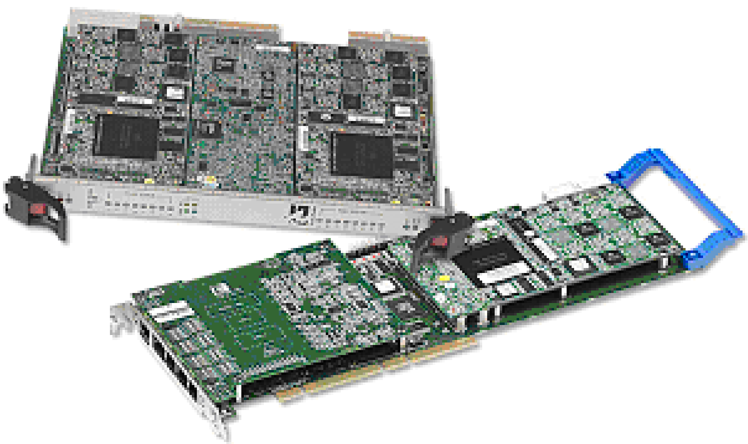
AudioCodes™ Mediant™ Digital Media Gateways & TrunkPack™ Boards

SIP

Mediant 2000
TP-1610 & TP-260/UNI Boards

User's Manual Version 5.0



Document #: LTRT-68805 **October 2006**

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Notice

This document describes the release of the AudioCodes Mediant 2000 SIP gateway, TP-1610 SIP cPCI board, and TP-260 SIP PCI board.
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Date Published: Oct-9-2006 Date Printed: Oct-12-2006

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Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used. Only industry-standard terms are used throughout this manual. Hexadecimal notation is indicated by 0x preceding the number.

Related Documentation

Document #	Manual Name
LTRT-690xx (e.g., LTRT-69001)	Mediant 3000 & Mediant 2000 & TP Series SIP Digital Gateways Release Notes
LTRT-701xx	Mediant 2000 Fast Track Guide
LTRT-665xx	CPE Configuration Guide for IP Voice Mail

Warning: The Mediant 2000 is supplied as a sealed unit and must only be serviced by qualified service personnel.

Note: Where 'network' appears in this manual, it means Local Area Network (LAN), Wide Area Network (WAN), etc. accessed via the gateway's Ethernet interface.

1 Overview

1.1 Introduction

This document provides you with the information on installation, configuration and operation of the Mediant 2000 SIP gateway, TP-1610 SIP cPCI board and TP-260 SIP PCI board. As these products have similar functionality (with the exception of their physical layout and the number of trunks), they are collectively referred to throughout this manual (except for in hardware-related sections) as the *gateway*.

1.2 Mediant 2000 Overview

The Mediant 2000 SIP Voice over IP (VoIP) gateway enables voice, fax, and data traffic to be sent over the same IP network. The Mediant 2000 provides excellent voice quality and optimized packet voice streaming over IP networks.

The Mediant 2000 uses the award-winning, field-proven Digital Signal Processing (DSP) voice compression technology used in other TrunkPack™ series products.

The Mediant 2000 incorporates 1, 2, 4, 8 or 16 E1 or T1 spans for connection, directly to Public Switched Telephone Network (PSTN) / Private Branch Exchange (PBX) telephony trunks, and includes one or two 10/100 Base-TX Ethernet ports for connection to the network.

The Mediant 2000 supports up to 480 simultaneous VoIP or Fax over IP (FoIP) calls, supporting various Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) protocols such as EuroISDN, North American NI2, Lucent™ 4/5ESS, Nortel™ DMS100 and others. In addition, it supports different variants of Channel Associated Signaling (CAS) protocols for E1 and T1 spans, including MFC R2, E&M immediate start, E&M delay dial/start, loop start and ground start.

The Mediant 2000 gateway, best suited for large and medium-sized VoIP applications, is a compact device, comprising a 19-inch 1U chassis with optional dual AC or single DC power supplies.

The deployment architecture can include several Mediant 2000 gateways in branch or departmental offices, connected to local PBXs. Call routing is performed by the gateways themselves or by SIP Proxy(s).

The Mediant 2000 gateway enables users to make low cost long distance or international telephone/fax calls between distributed company offices, using their existing telephones/fax. These calls are routed over the existing network ensuring that voice traffic uses minimum bandwidth.

The Mediant 2000 can also route calls over the network using SIP signaling protocol, enabling the deployment of Voice over Packet solutions in environments where access is enabled to PSTN subscribers by using a trunking media gateway. This provides the ability to transmit voice and telephony signals between a packet network and a TDM network. Routing of the calls from the PSTN to a SIP service node (e.g., Call Center) is performed by the Mediant 2000 internal routing feature or by a SIP Proxy.

Note: The Mediant 2000 is offered as a 1-module (up to 240 channels or 8 trunk spans) or 2-module (for 480 channels or 16 trunk spans only) platform. The latter configuration supports two TrunkPack modules, each having its own IP address. Configuration instructions in this document relate to the Mediant 2000 as a 1-module platform and must be repeated for the second module as well.

1.3 TP-1610 Overview

The TP-1610 cPCI VoIP media gateway board, based on dual TPM-1100 PMC Modules, is a complete SIP-compliant 'two media gateways on a board', delivering cost-effective solution in a convenient cPCI form-factor.

The TP-1610 is an ideal solution for SIP trunking gateways and integrated media gateways for IP-PBXs and all-in-one communication servers. The board is designed for enterprise or carrier applications. The TP-1610 provides up to 480 simultaneous ports for voice, fax or data for VoIP media gateway applications providing excellent voice quality and optimized packet voice streaming over IP networks. Employing SIP as a control protocol, the TP-1610 enables vendors and System Integrators (SIs) short time-to-market and reliable cost-effective deployment of next-generation networks.

The TP-1610 matches the density requirements for small to medium locations, while meeting Network Service Providers' (NSP) demands for scalability. The TP-1610, scales from no trunk spans to 16 E1/T1/J1 spans in a single cPCI slot and provides an excellent gateway solution for enterprise applications as well as carrier locations.

One or two packet processors (depending on the board's capacity) handle packet-streaming functions through two, redundant integral 10/100 Base-TX interfaces. Each processor implements the industry-standard RTP/RTCP packet-streaming protocol, advanced adaptive jitter buffer management, and T.38 fax relay over IP.

The TP-1610 supports various ISDN PRI protocols such as EuroISDN, North American NI2, Lucent™ 4/5ESS, Nortel™ DMS100 and others. In addition, it supports different variants of CAS protocols for E1 and T1 spans, including MFC R2, E&M immediate start, E&M delay dial / start, loop start and ground start.

The TP-1610 enables the deployment of 'Voice over Packet' solutions in environments where access is enabled to PSTN subscribers by using a trunking media gateway. This provides the ability to transmit voice and telephony signals between a packet network and a TDM network. Routing of the calls from the PSTN to a SIP service node (e.g., Call Center) is performed by the TP-1610 internal routing feature or by a SIP Proxy.

Enabling accelerated design cycles with higher density and reduced costs, the TP-1610 is an ideal building block for scalable, reliable VoIP solutions. With the TP-1610's comprehensive feature set, customers can quickly design a wide range of solutions for PSTN and VoIP networks.

Note: The TP-1610 is offered as a 1-module (up to 240 channels or 8 trunk spans) or 2-module (for 480 channels or 16 trunk spans only) platform. The latter configuration supports two TrunkPack modules, each having its own IP address. Configuration instructions in this document relate to the TP-1610 as a 1-module platform and must be repeated for the second module as well.

1.4 TP-260 Overview

The TP-260 SIP PCI VoIP media gateway board is a complete SIP-compliant ‘media gateway on a board’, delivering cost-effective solution in a convenient PCI form-factor. This unique stand-alone PCI media gateway operates independently and relies on the host PCI only for its power. The TP-260 communicates to applications via SIP using an on-board Ethernet interface. Using a special standards-based approach eliminates host PC device drivers and operation system dependencies, seamlessly connecting existing PSTN-based systems to support VoIP.

The TP-260 is an ideal solution for SIP trunking gateways and integrated media gateways for IP-PBXs and all-in-one communication servers. The board is designed for enterprise applications or for smaller to medium PC-based systems. The TP-260 provides up to 240 simultaneous ports for voice, fax or data for VoIP media gateway applications providing excellent voice quality and optimized packet voice streaming over IP networks. Employing SIP as a control protocol, the TP-260 enables System Integrators short time-to-market and reliable cost-effective deployment of next-generation networks. The TP-260 utilizes the TPM-1100 PMC module, which is based on the VolPerfect™ architecture, AudioCodes' underlying core media gateway technology.

The TP-260 matches the density requirements for small to medium locations, while meeting NSP’s demands for scalability. The TP-260 stand-alone VoIP gateway on a board, scales from 1 to 8 E1/T1/J1 spans in a single PCI slot and provides an excellent gateway solution for enterprise applications as well as carrier locations.

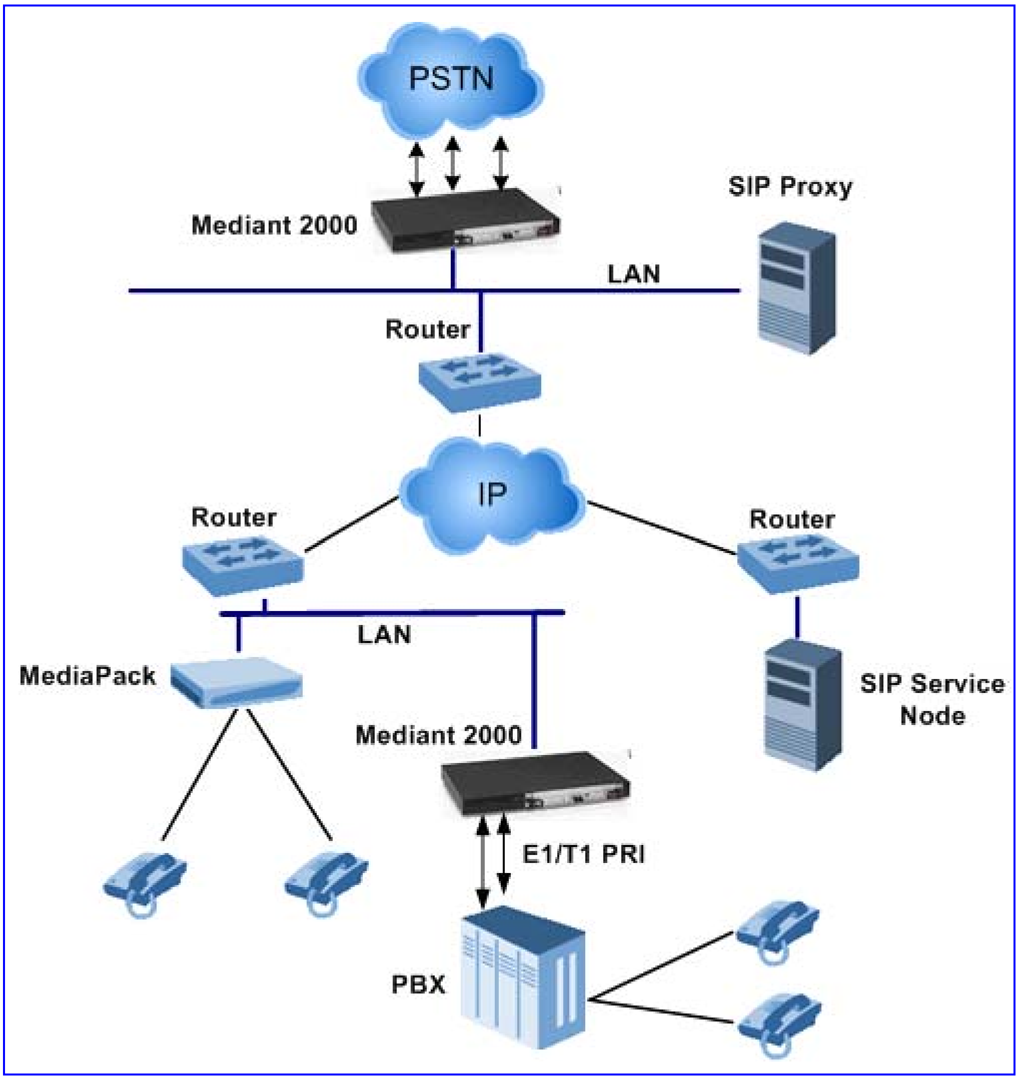
The TP-260 supports various ISDN PRI protocols such as EuroISDN, North American NI2, Lucent™ 4/5ESS, Nortel™ DMS100 and others. In addition, it supports different variants of CAS protocols for E1 and T1 spans, including MFC R2, E&M immediate start, E&M delay dial / start, loop start and ground start.

The TP-260 enables the deployment of ‘Voice over Packet’ solutions in environments where access is enabled to PSTN subscribers by using a trunking media gateway. This provides the ability to transmit voice and telephony signals between a packet network and a TDM network. Routing of the calls from the PSTN to a SIP service node (e.g., Call Center) is performed by the TP-260 internal routing feature or by a SIP Proxy.

Enabling accelerated design cycles with higher density and reduced costs, the TP-260 is an ideal building block for scalable, reliable VoIP solutions. With the TP-260 comprehensive feature set, customers can quickly design a wide range of solutions for PSTN and VoIP networks.

Figure 1-1 below illustrates typical Mediant 2000 / TP-1610 / TP-260 gateway applications over VoIP network.

Figure 1-1: Typical Mediant 2000 / TP-1610 / TP-260 Gateway Application



1.5 SIP Overview

SIP is an application-layer control (signaling) protocol used on the gateway for creating, modifying, and terminating sessions with one or more participants. These sessions can include Internet telephone calls, media announcements and conferences.

SIP invitations are used to create sessions and carry session descriptions that enable participants to agree on a set of compatible media types. SIP uses elements called proxy servers to help route requests to the user's current location, authenticate and authorize users for services, implement provider call-routing policies and provide features to users.

SIP also provides a registration function that enables users to upload their current locations for use by proxy servers. SIP, on the gateway, complies with the IETF (Internet Engineering Task Force) RFC 3261 (refer to www.ietf.org/rfc/rfc3261.txt?number=3261).

1.6 Features

This section provides a high-level overview of some of the many gateway supported features.

1.6.1 General Features

- Compliant with SIP (RFC 3261).
- Supported coders: G.711 A-law, G.711 μ-law, G.723.1, G.726, G.729, NetCoder, EVRC, AMR, Transparent, GSM Full-Rate, Microsoft GSM and GSM EFR.
  When EVRC (Enhanced Variable Rate Codec), AMR (Adaptive Multi-Rate) and GSM EFR are used, the number of available gateway channels is reduced (refer to the Mediant 3000 & Mediant 2000 & TP Series SIP Digital Release Notes).
- Supports negotiation of dynamic payload types.
- Supports reception and DNS resolution of FQDNs received in SDP.
- T.38 fax with superior performance (handling a round-trip delay of up to nine seconds).
- Echo Canceler (with up to 128 msec tail length), Jitter Buffer, Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) support.
- Silence suppression with Comfort Noise Generation.
- Web management for easy configuration and installation.
- EMS for comprehensive management operations (FCAPS).
- Simple Network Management Protocol (SNMP) and Syslog support.
- SMDI support for Voice Mail applications (Mediant 2000 only).
- ThroughPacket™ proprietary feature that aggregates payloads from several channels into a single IP packet to reduce bandwidth overhead.
- Supports load balancing with proxy.
- Can be integrated into a Multiple IPs and a VLAN-aware environment.
- Capable of automatically updating its firmware version and configuration.
- Secured Web access (HTTPS) and Telnet access using SSL / TLS.
- IPSec and IKE protocols are used in conjunction to provide security for control (e.g., SIP) and management (e.g., SNMP and Web) protocols.
- Secured RTP (SRTP) according to RFC 3711, used to encrypt RTP and RTCP transport.

1.6.2 PSTN-to-SIP Interworking Features

The gateway performs interworking between ISDN and CAS via E1/T1/J1 digital spans and SIP IETF signaling protocol. 16 E1, T1 or J1 spans are supported (480 channels) in a two modules gateway.

The gateway supports various ISDN PRI protocols such as EuroISDN, North American NI2, Lucent 4/5ESS, Nortel DMS100, Meridian 1 DMS100, Japan J1, as well as QSIG. PRI support includes User Termination or Network Termination side. ISDN-PRI protocols can be defined on an E1/T1 basis (i.e., different variants of PRI are allowed on different E1/T1 spans).

In addition, it supports numerous variants of CAS protocols for E1 and T1 spans, including MFC R2, E&M wink start, E&M immediate start, E&M delay dial/start, loop-start, and ground start. CAS protocols can be defined on an E1/T1 basis (i.e., different variants of CAS are allowed on different E1/T1 spans).

PSTN to SIP and SIP to PSTN Called number can be optionally modified according to rules that are defined in gateway *ini* file.

The supported interworking features include the following:

- Definition and use of Trunk Groups for routing IP→PSTN calls.
- B-channel negotiation for PRI spans.
- ISDN Non Facility Associated Signaling (NFAS).
- PRI to SIP interworking according to <draft-ietf-sipping-qsig2sip-04.txt>.
- PRI to SIP Interworking of Q.931 Display (Calling name) information element.
- PRI (NI-2, 5ESS) to SIP interworking of Calling Name using Facility IE in Setup and Facility messages.
- Configuration of Numbering Plan and Type for IP→ISDN calls.
- Interworking and flexible mapping of PSTN to SIP release causes.
- Interworking of ISDN redirect number to SIP diversion header (according to IETF <draft-levy-sip-diversion-05.txt>).
- Optional change of redirect number to called number for ISDN→ IP calls.
- Interworking of ISDN calling line Presentation & Screening indicators using RPID header <draft-ietf-sip-privacy-04.txt>.
- Interworking of Q.931 Called and Calling Number Type and Number Plan values using the RPID header.
- Interworking of Calling and Called Subaddress values for SIP→ISDN calls.
- Supports ISDN en-block or overlap dialing for incoming Tel→IP calls.
- Supports a digit map pattern to reduce the dialing period when Overlap dialing is used.
- Supports routing of IP→Tel calls to predefined trunk groups.
- Supports a configurable channel select mode per trunk group.
- Supports various number manipulation rules for IP→Tel and Tel→IP, called and calling numbers.
- Option to configure ISDN Transfer Capability per trunk.
- Supports QSIG messages tunneling over SIP according to <draft-elwell-sipping-qsig-tunnel-03>.
- Supports ISDN PRI Setup and Facility messages tunneling over SIP INVITE and INFO messages.
- Interworking of Redirect Number for QSIG□SIP calls.
- Supports QSIG Call Re-Route.
- Supports QSIG MWI Notifications.

1.6.3 Supported SIP Features

- Reliable User Datagram Protocol (UDP) transport, with retransmissions.
- Transmission Control Protocol (TCP) Transport layer.
- SIPS using TLS.
- T.38 real time fax (using SIP).
  Note: If the remote side includes the fax maximum rate parameter in the SDP body of the INVITE message, the gateway returns the same rate in the response SDP.
- Works with Proxy or without Proxy, using an internal routing table.
- Fallback to internal routing table if Proxy is not responding.
- Supports up to four Proxy servers. If the primary Proxy fails, the gateway automatically switches to a redundant Proxy.
- Supports domain name resolving using DNS SRV records for Proxy, Registrar and domain names that appear in the Contact and Record-Route headers.

- Proxy and Registrar Authentication (handling 401 and 407 responses) using Basic or Digest methods. Accepted challenges are kept for future requests to reduce the network traffic.
- Single gateway Registration or multiple Registration of all gateway endpoints.
- Supported methods: INVITE, CANCEL, BYE, ACK, REGISTER, OPTIONS, INFO, REFER, UPDATE, NOTIFY, PRACK and SUBSCRIBE.
- Modifying connection parameters for an already established call (re-INVITE).
- Working with a Redirect server and handling 3xx responses.
- Early Media (supporting 183 Session Progress).
- PRACK reliable provisional responses (RFC 3262).
- Call Hold and Transfer Supplementary services using REFER, Refer-To, Referred-By, Replaces and NOTIFY messages.
- Supports RFC 3711, Secured RTP and Key Exchange according to <draft-ietf-mmusic-sdescriptions-12>.
- Supports RFC 3489, Simple Traversal of UDP Through NATs (STUN).
- Supports RFC 3327, Adding ‘Path’ to Supported header.
- Supports RFC 3581, Symmetric Response Routing.
- Supports RFC 3326, Reason header.
- Supports RFC 4028, Session Timers in SIP.
- Locating SIP Servers (RFC 3263).
- An Offer/Answer Model with Session Description Protocol (SDP) (RFC 3264).
- Supports network asserted identity and privacy (RFC 3325 and RFC 3323).
- Supports Tel URI (Uniform Resource Identifier) according to RFC 2806 bis.
- Remote party ID <draft-ietf-sip-privacy-04.txt>.
- Supports obtaining Proxy Domain Name(s) from DHCP (Dynamic Host Control Protocol) according to RFC 3361.
- RFC 2833 Relay for Dual Tone Multi Frequency (DTMF) digits, including payload type negotiation.
- DTMF out-of-band transfer using:
  - INFO method <draft-choudhuri-sip-info-digit-00.txt>
  - INFO method, compatible with Cisco gateways
  - NOTIFY method <draft-mahy-sipping-signaled-digits-01.txt>.
- SIP URI: sip:”phone number”@IP address (such as 1225556@10.1.2.4, where “122556” is the phone number of the source or destination) or sip:”phone_number”@”domain name”, such as 122556@myproxy.com. Note that the SIP URI host name can be configured differently per called number.
- Supports RFC 4040, RTP payload format for a 64 kbit/s transparent data.
- Can negotiate coder from a list of given coders.
- Responds to OPTIONS messages both outside a SIP dialog and in mid-call. Generates SIP OPTIONS messages as Proxy keep-alive mechanism.
- Representing trunk groups in tel/sip Uniform Resource Identifiers (URIs) according to <draft-ietf-iptel-trunk-group-04>.
- The number of total and free channels is published in a 200 OK response to an OPTIONS request. The gateway uses the X-Resource header in the following format: ‘X-Resource: telchs=100/240;mediachs=0/0’,
Where ‘telchs’ specifies the number of free tel channels / total tel channels.

For more updated information on the gateway’s supported features, refer to the latest Mediant & TP Series SIP Digital Gateways Release Notes.

Reader's Notes

2 Physical Description

This section provides detailed information on the hardware components, the location and functionality of the LEDs, buttons and connectors of the following products:

- Mediant 2000 (refer to Section 2.1 below).
- TP-1610 (refer to Section 2.2 on page 29).
- TP-260 (refer to Section 2.3 on page 32).

2.1 Mediant 2000 Physical Description

The Mediant 2000 (shown in Figure 2-1 below) comprises the following components:

- A 19-inch 1U high rack mount chassis (refer to Section 2.1.1 on page 28).
- A single compactPCI™ TP-1610 board (refer to Section 2.2 on page 29).
- A single TP-1610 Rear Transition Module (RTM) (refer to Section 2.2.2 on page 31).
- A single available cPCI slot for an optional third-party CPU board (refer to Section 2.1.2 on page 31).

Figure 2-1: Mediant 2000 Front View

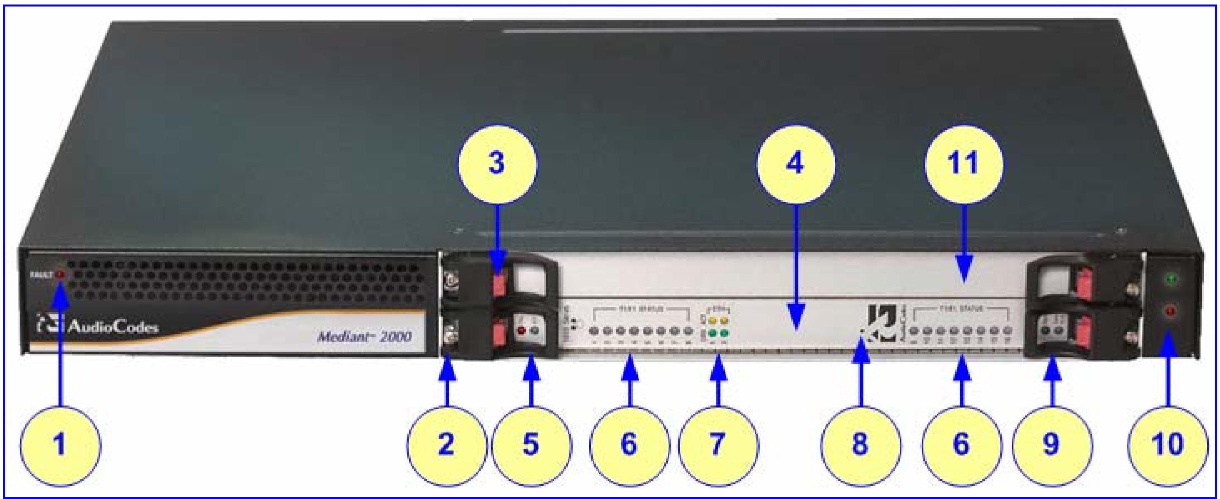


Table 2-1: Mediant 2000 Front View Component Descriptions

Item #	Label	Component Description
1	FAULT	Dual AC Power LED.
2		cPCI board locking screws.
3		cPCI latches.
4		TP-1610 cPCI board, 16-trunk configuration.
5		Status LED Indicators.
6	T1/E1 STATUS	E1/T1 Trunk Status LED Indicators.
7	ETH	Ethernet LED Indicators.
8		Reset button.
9		cPCI LED Indicators.
10		Power and Fan LEDs
11		An available cPCI slot for an optional third-party CPU board.

2.1.1 The Mediant 2000 Chassis

The Mediant 2000 chassis is an industrial platform, 19” wide, 1U high and 12” deep that houses the TP-1610 board in its front cage, slot #1 (the lower slot) and the TP-1610 RTM in its rear cage, slot #1 (the lower slot).

Slot # 2 in the Mediant 2000 chassis’ front and rear cages can optionally be used by customers for a CPU board.

Refer to Table 2-2 for detailed description of the chassis’ LED indicators.

Table 2-2: Chassis LED Indicators

Location	Color	Function
Right side of front panel	Green	The power is on.
Right side of front panel	Red	Fan failure - indicates that any of the internal fans has significantly reduced its speed or has stopped.
Left side of front panel	Red	Power supply failure - indicates that one of the two AC redundant power supplies is faulty or disconnected from the AC/mains outlet. (This LED is only relevant for the dual AC power supply).

2.1.1.1 Power Supply

The Mediant 2000 power supply is available in three configuration options:

- Single universal 100-240 VAC 1 A max, 50-60 Hz.
- Dual-redundant 100-240 VAC 1.5 A max, 50-60 Hz.
- -48 VDC power supply suitable for field wiring applications.

2.1.2 Optional CPU Board

The Mediant 2000 provides an optional second cPCI slot that can be optionally used for customer’s CPU board. This CPU board can be used for general applications such as a Gatekeeper, Softswitch, Application Server or other. The following CPU boards were tested for compliancy with the Mediant 2000 chassis:

- Sun™: CP2080 + PMC-233 (Ramix™ disk on board) + Rear Transition Module (RTM).
- Intel™ ZT5515B-1A with 40GB on-board disk plus RTM (ZT4807).

For details on removing / inserting the optional CPU board, refer to the directions accompanying it.

2.2 TP-1610 Physical Description

The TP-1610 (shown in Figure 2-2) is a high-density, hot-swappable, cPCI resource board with a capacity of up to 480 ports, supporting all necessary functions for voice, data and fax streaming over IP networks. The TP-1610 is composed of one or two identical media gateway modules: Gateway-1 and Gateway-2, each containing 240 DSP channels. These media gateways are fully independent, each gateway having its own MAC (Media Access Control) and IP addresses and LED indicators. The TP-1610 board is supplied with a rear I/O configuration in which both PSTN trunks and Ethernet interface are located on a passive rear I/O module (for information on the RTM, refer to Section 2.2.2 on page 31).

Figure 2-2: Front and Upper View of the TP-1610 cPCI Board

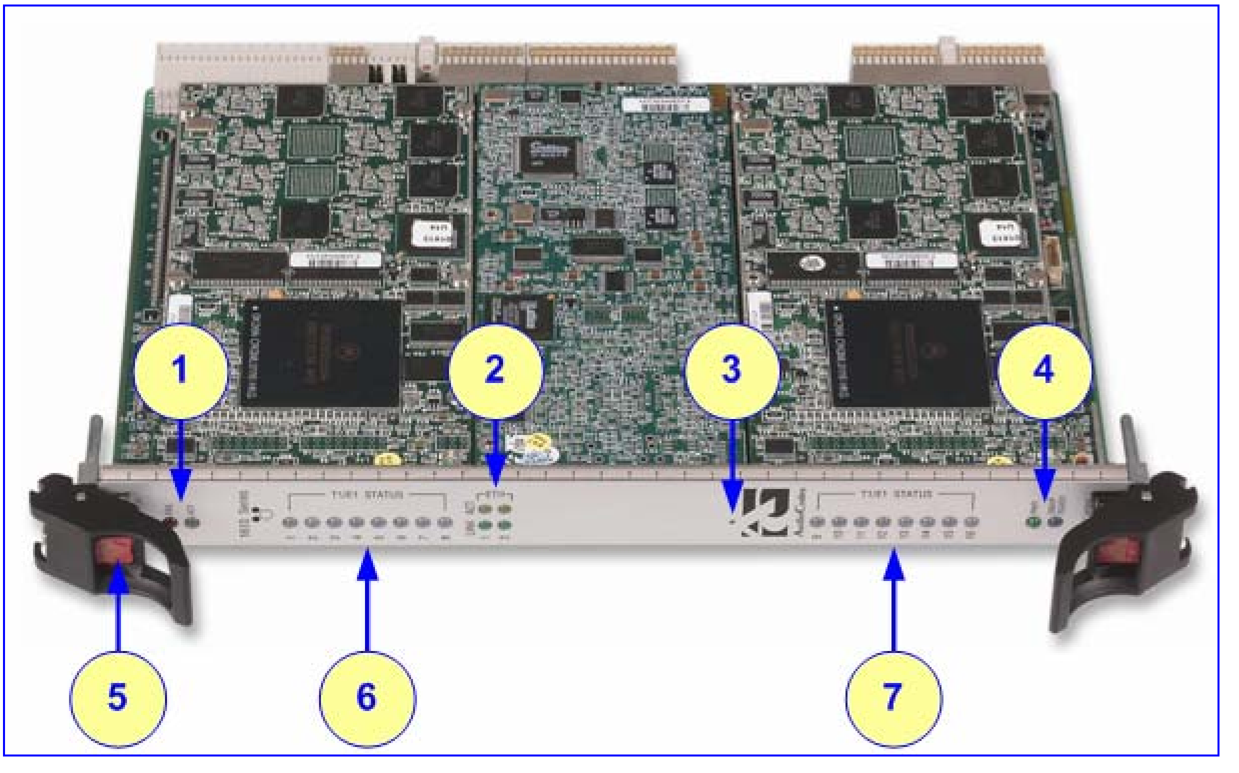


Table 2-3: Front and Upper View of the TP-1610 cPCI Board Component Descriptions

Item #	Label	Component Description
1		Status LEDs
2	**ETH**	Ethernet LEDs
3		Reset button
4		cPCI LEDs
5		cPCI Latch
6	**T1 / E1 STATUS**	T1/E1 Trunk Status LEDs (for each of trunks 1 to 8)
7	**T1 / E1 STATUS**	T1/E1 Trunk Status LEDs (for each of trunks 9 to 16)

2.2.1 TP-1610 Front Panel LED Indicators

The functionality of the front panel LEDs for the TP-1610 is described in the following four tables and illustrated in Figure 2-2 on page 29. Note that there is a choice of front panels according to the number of channels.

Table 2-4: Status LED Indicators

Label	LED Color	LED Function
FAIL	Red	Indicates gateway failure (fatal error)
	--	Off indicates normal functioning
ACT	Green	Gateway initialization sequence terminated OK
	Yellow	N/A

Table 2-5: E1/T1 Trunk Status LED Indicators

Label	LED Color	Signal Description
T1/E1 Status 1 to 8 and T1/E1 Status 9 to 16	Green	Trunk is synchronized (normal operation)
	Red	Loss due to any of the following 4 signals: ▪ **LOS** Loss of Signal ▪ **LOF (**Loss of Frame) ▪ **AIS** (Alarm Indication Signal -- 'Blue alarm') ▪ **RAI** (Remote Alarm Indication -- 'Yellow alarm')

Note: On the front panel 16 LEDs are provided for 16-span units and 8 LEDs are provided for 1-span, 2-span, 4-span, and 8-span units. In the case of 1-span, 2-span and 4-span units, the extra LEDs are unused.

Table 2-6: Ethernet LED Indicators

Label	LED Color	LED Function
LINK	Green	Link all OK
ACT	Yellow	Transmit / receive activity

Table 2-7: cPCI LED Indicators

Label	LED Color	LED Function
PWR	Green	Power is supplied to the board
SWAP READY	Blue	The cPCI board can now be removed.
		The cPCI board was inserted successfully. For detailed information on inserting / removing the TP-1610 board, refer to Section 3.2.3 on page 44.

During correct operation, the ACT LED is lit green, the FAIL LED is off. Changing of the FAIL LED to red indicates a failure.

2.2.2 Rear Transition Module

The RTM includes PSTN trunks, an Ethernet interface, and an optional RS-232 connector (available only on the 1, 2 and 4-span configuration).

The Ethernet interface features dual 10/100 Base-TX, RJ-45 shielded connectors for (an active / standby) redundancy scheme providing protection against the event of a failure.

The PSTN interface is available in 1-span, 2-span, 4-span, 8-span, or 16-span rear panels.

Rear panel with two 50-pin female Telco connectors (DDK 57AE-40500-21D) (shown in Figure 2-3) is required for a gateway equipped with 16 E1/T1 spans. Rear panel with RJ-48c connectors (shown in Figure 2-4) is required for a gateway equipped with 1, 2, 4, or 8 E1/T1 spans. The physical difference between the 1-Span, 2-Span and 4-Span RTMs, and the 8-span RTM is that the RJ-48c ports are depopulated correspondingly.

Figure 2-3: Rear Panel with two 50-pin Connectors for 16 Trunks

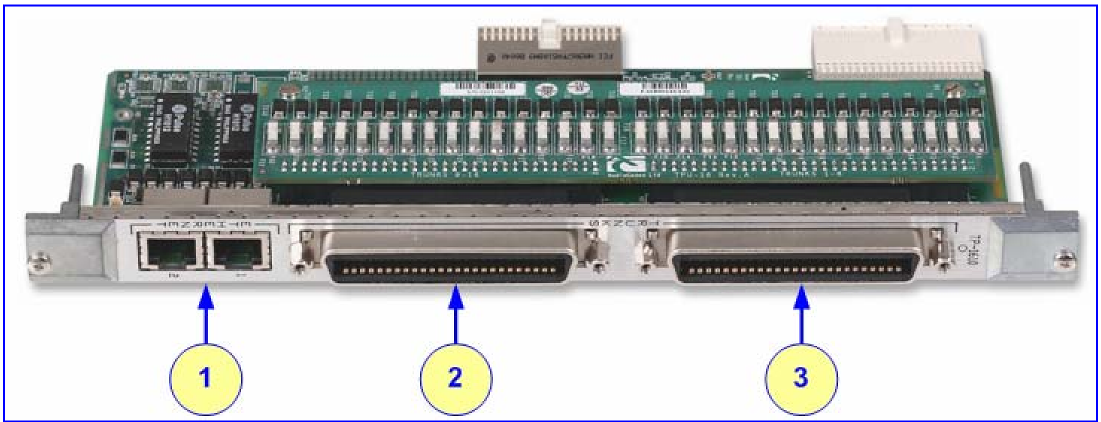


Table 2-8: Rear Panel with two 50-pin Connectors for 16 Trunks Component Descriptions

Item #	Label	Component Description
1	**ETHERNET**	2 Ethernet Ports. 2 RJ-45 network connectors.
2	**TRUNKS**	E1/T1 trunks 9 to 16. 50-pin female Telco connector.
3	**TRUNKS**	E1/T1 trunks 1 to 8. 50-pin female Telco connector.

Figure 2-4: Rear Panel with 8 RJ-48c Connectors for 8 Trunks

Table 2-9: Rear Panel with 8 RJ-48c Connectors for 8 Trunks Component Descriptions

Item #	Label	Component Description
1	**ETHERNET**	2 Ethernet Ports. 2 RJ-45 network connectors
2	**TRUNKS**	8 E1/T-1 Spans. 8 RJ-48c trunk connectors

2.3 TP-260 Physical Description

The TP-260 board is a fully ‘Plug and Play’ device. The PC’s boot-up sequence determines its I/O addresses and interrupts.

Figure 2-5: The TP-260 Board

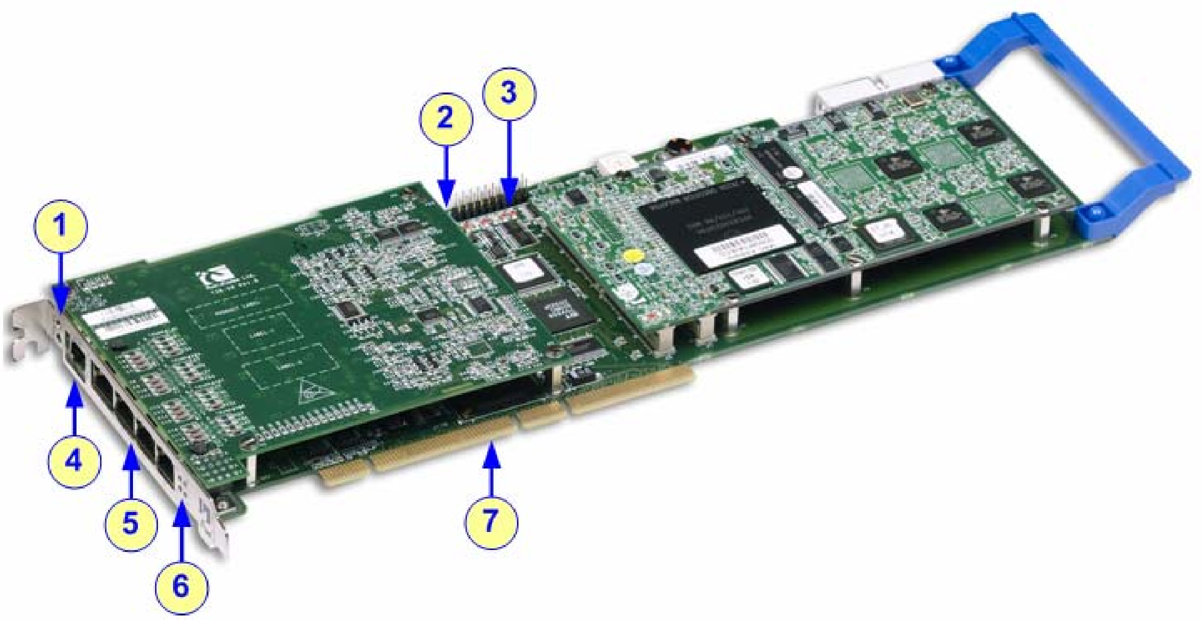
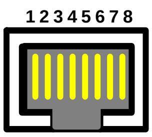


Table 2-10: TP-260 Component Descriptions

Item #	Component Description
1	Ethernet LEDs (refer to Table 2-11 on page 33)
2	Reset button
3	Internally-located base board LEDs (refer to Table 2-13 on page 33)
4	Ethernet RJ-45 connector (for pinout, refer to Figure 2-6 below)
5	Four T1/E1 RJ-48c trunk connectors (for pinout, refer to Figure 2-7 on page 32)
6	E1/T/J1 LEDs (refer to Table 2-12 on page 33)
7	Universal PCI, 32/64 bit, 33/66 MHz, and 3.3/5 V.

Figure 2-6: Pinout of the RJ-45 Connector

RJ-45 LAN Connector and Pinout

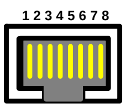


1 = Tx+
2 = Tx-
3 = Rx+
6 = Rx-

4, 5, 7, 8
not connected

Figure 2-7: Pinout of the RJ-48c Trunk Connectors

RJ-48c Connector and Pinout



1 = Rx RING
2 = Rx TIP
4 = Tx RING
5 = Tx TIP

3, 6, 7, 8
not connected

body = shield

2.3.1 TP-260 LEDs

Refer to Table 2-11 through to Table 2-13 for LEDs.

Table 2-11: Ethernet LEDs

Label	LED Color	Function
Rx	**Yellow**	Receiving data
Link	**Green**	Ethernet connection is ON (Link)

Table 2-12: E1/T/J1 LEDs on the Front Panel (Bracket)

Label	Color	Signal Description
Trunk Status 1 to 8	**Green**	Trunk is synchronized (normal operation)
	Red	Loss due to one of the following signals: • **LOS** (Loss of Signal) • **LFA** (Loss of Frame Alignment) • **AIS** (Alarm Indication Signal -- 'Blue alarm') • **RAI** (Remote Alarm Indication -- 'Yellow alarm')

Table 2-13: Internally-Located Base Board LEDs

LED	Name	Color	Signal Description
LD1	COL	**Red**	Link collision. The LED toggles when there is a collision in the half-duplex operation.
LD2	SPEED	**Orange**	Link speed 10/100 Base-TX. The LED is ON for 100 Mbps and OFF for 10 Mbps.
LD3	DUPLEX	**Red**	Link half-duplex or full-duplex. The LED is ON for full-duplex and OFF for half-duplex.
LD4	TX	**Orange**	Link Transmit. When the PHY transmits, the LED toggles.
LD5	RS-232	**Red**	Internal use only.
LD6	FAIL	**Red**	Indication from the TPM-1100.
LD7	CLK40M	**Red**	When the LED toggles, the CLK40M for the PCI controller is active.

Reader's Notes

3 Installation

This section provides detailed information on the installation procedures for the following products:

- Mediant 2000 (refer to Section 3.1 below).
- TP-1610 (refer to Section 3.2 on page 44).
- TP-260 (refer to Section 3.3 on page 46).

For information on how to start using the gateway, refer to Chapter 4 on page 49.

Caution Electrical Shock

The equipment must only be installed or serviced by qualified service personnel.

3.1 Installing the Mediant 2000

➢ To install the Mediant 2000, take these 4 steps:

1. Unpack the Mediant 2000 (refer to Section 3.1.1 below).
2. Check the package contents (refer to Section 3.1.2 below).
3. Mount the Mediant 2000 (refer to Section 3.1.3 on page 36).
4. Cable the Mediant 2000 (refer to Section 3.1.4 on page 38).

After powering-up the Mediant 2000, the **Ready** and **LAN** LEDs on the front panel turn to green (after a self-testing period of about 3 minutes). Any malfunction changes the **Ready** LED to red (refer to Section 2.2.1 on page 30 for details on the Mediant 2000 LEDs).

When you have completed the above relevant sections you are then ready to start configuring the gateway (Chapter 4 on page 49).

3.1.1 Unpacking

➢ To unpack the Mediant 2000, take these 6 steps:

1. Open the carton and remove packing materials.
2. Remove the Mediant 2000 gateway from the carton.
3. Check that there is no equipment damage.
4. Check, retain and process any documents.
5. Notify AudioCodes or your local supplier of any damage or discrepancies.
6. Retain any diskettes or CDs.

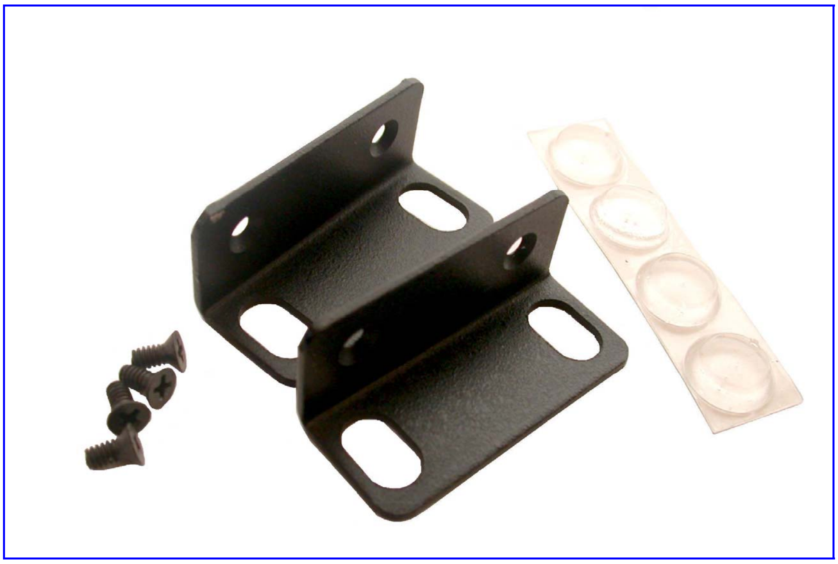
3.1.2 Package Contents

Ensure that in addition to the Mediant 2000, the package contains:

- For the dual AC power supply version two AC power cables are supplied; for the single AC power supply version one AC power cable is supplied.
- For the DC power supply version, one connectorized DC power cable (crimp connection type) and one DC adaptor (screw connection type) connected to the rear panel of the Mediant 2000 are supplied; use only one type.
- CD (software and documentation).

- Small plastic bag containing (refer to Figure 3-1):
  - Two brackets and four bracket-to-device screws for 19-inch rack installation option.
  - Four anti-slide bumpers for desktop / shelf installation option.
- The Mediant 2000 Fast Track Installation Guide.

Figure 3-1: 19-inch Rack & Desktop Accessories



3.1.3 Mounting the Mediant 2000

The Mediant 2000 can be mounted on a desktop, or installed in a standard 19-inch rack. Refer to Section 3.1.4 on page 38 for cabling the Mediant 2000.

3.1.3.1 Mounting the Mediant 2000 on a Desktop

No brackets are required. Optionally, attach the four (supplied) anti-slide bumpers to the base of the Mediant 2000 and place it on the desktop in the position you require.

3.1.3.2 Installing the Mediant 2000 in a 19-inch Rack

Users can install the device in a standard 19-inch rack either by placing the device on a shelf preinstalled in the rack (preferred method), or by attaching the device directly to the rack's frame via integral brackets.

Before rack mounting the chassis, attach the two (supplied) brackets to the front sides of the device (refer to Figure 3-2).

➢ To attach the two front side brackets, take these 3 steps:

1. Remove the 2 screws nearest the front panel on either side of the device.
2. Align a bracket over 2 holes on one side (so that the bracket's larger holes face front) and with the 2 supplied replacement screws, screw in the bracket.
3. Perform the same procedure on the other side.

Figure 3-2: Mediant 2000 Front View with 19-inch Rack Mount Brackets

Rack Mount Safety Instructions (UL)

When installing the chassis in a rack, be sure to implement the following Safety instructions recommended by Underwriters Laboratories:

- **Elevated Operating Ambient** - If installed in a closed or multi-unit rack assembly, the operating ambient temperature of the rack environment may be greater than room ambient. Therefore, consideration should be given to installing the equipment in an environment compatible with the maximum ambient temperature (Tma) specified by the manufacturer.
- **Reduced Air Flow** - Installation of the equipment in a rack should be such that the amount of air flow required for safe operation on the equipment is not compromised.
- **Mechanical Loading** - Mounting of the equipment in the rack should be such that a hazardous condition is not **achieved** due to uneven mechanical loading.
- **Circuit Overloading** - Consideration should be given to the connection of the equipment to the supply circuit **and** the effect that overloading of the circuits might have on overcurrent protection and supply wiring. Appropriate consideration of equipment nameplate ratings should be used when addressing this concern.
- **Reliable Earthing** - Reliable earthing of rack-mounted equipment should be maintained. Particular attention should be given to supply connections other than direct connections to the branch circuit (e.g., use of power strips.)

➢ To attach the device to a 19-inch rack, take these 2 steps:

1. Position the device in your 19-inch rack and align the left-hand and right-hand bracket holes to holes (of your choosing) in the vertical tracks of the 19-inch rack.
2. Use standard 19-inch rack bolts (not provided) to fasten the device to the frame of the rack.

AudioCodes recommends using two additional (not supplied) rear mounting brackets to provide added support.

Note: Users assembling the rear brackets by themselves should note the following:

- The distance between the screws on each bracket is 26.5 mm.
- To attach the brackets, use 4-40 screws with a maximal box penetration length of 3.5 mm.

➢ To place the device on a 19-inch rack's shelf, take these 2 steps:

1. Place the device on the preinstalled shelf.
2. You're now recommended to take the optional steps of fastening the device to the frame of the rack (as described above) while it is placed on the shelf, so preventing it from sliding when inserting cables into connectors on the rear panel.

3.1.4 Cabling the Mediant 2000

Refer to Chapter 2 on page 27 for detailed information on the Mediant 2000 rear panel connectors and LEDs. Note that the Mediant 2000 is available in many *configurations*, i.e., AC or DC, in the 16-trunk, 8-trunk, 4-trunk, 2-trunk or 1-trunk device. The 16-trunk dual AC (Figure 3-3) and the 8-trunk DC (Figure 3-4) configurations are illustrated here *as representative* products.

Figure 3-3: Mediant 2000 Rear Panel Cabling (16 Trunks, Dual AC Power)

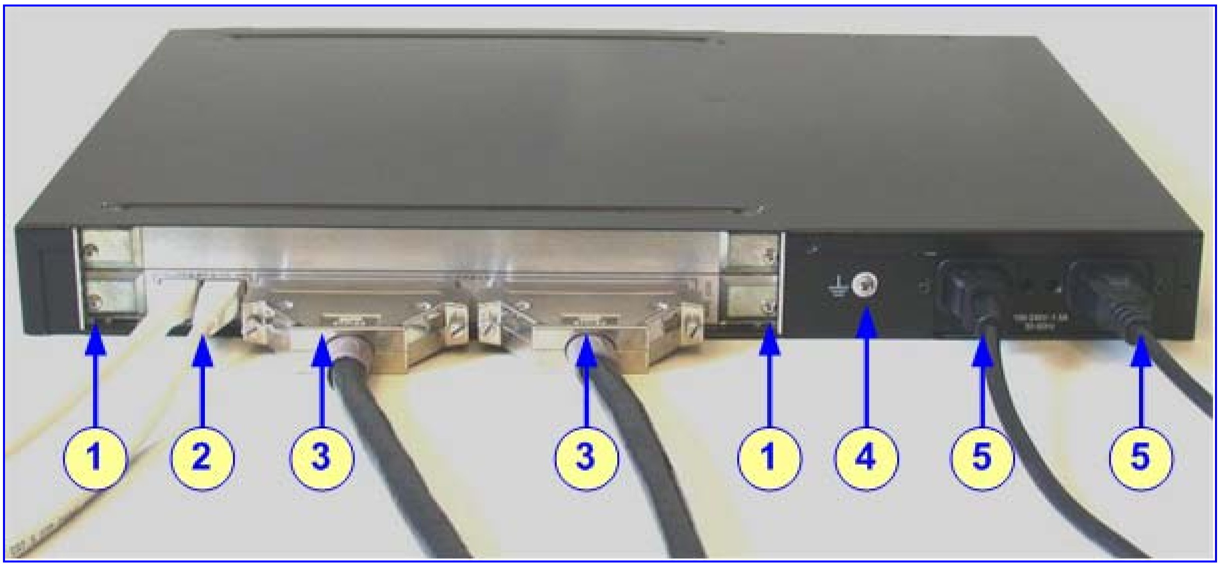


Table 3-1: Mediant 2000 Rear Panel Cabling (16 Trunks, Dual AC Power) Component Descriptions

Item #	Label	Component Description
1		RTM locking screws.
2	**ETHERNET**	Two Category 5 network cables, connected to the 2 Ethernet RJ-45 ports.
3	**TRUNKS**	Two 50-pin Telco connector cables (at least 26 AWG UTP), each supporting 8 trunks.
4	⏚	Protective earthing screw.
5	**100-240~1.5A**	Dual AC power cables.

Figure 3-4: Mediant 2000 Rear Panel Cabling (8 Trunks, DC Power)

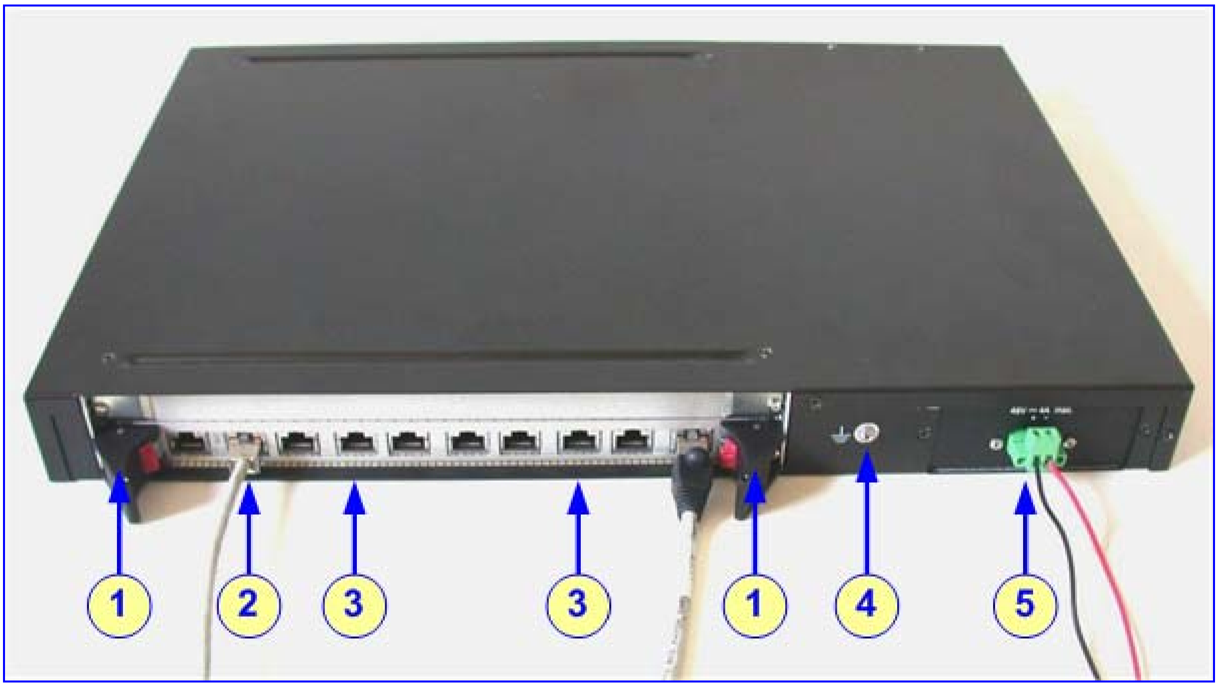


Table 3-2: Mediant 2000 Rear Panel Cabling (8 Trunks, DC Power) Component Descriptions

Item #	Label	Component Description
1		RTM latches.
2	**ETH**	A Category 5 network cable, connected to the Ethernet 1 RJ-45 port.
3	**PSTN**	8 RJ-48c ports, each supporting a trunk.
4	⏚	Protective earthing screw.
5	**48V 4A max**	2-pin connector for DC.

Electrical Earthing

The unit must be permanently connected to earth via the screw provided at the back on the unit. Use 14-16 AWG wire and a proper ring terminal for the earthing.

➢ To cable the Mediant 2000, take these 5 steps:

1. Permanently connect the device to a suitable earth with the protective earthing screw on the rear connector panel, using 14-16 AWG wire.
2. Connect the E1/T1 trunk interfaces (refer to Section 3.1.4.1 below).
3. Install the Ethernet connection (refer to Section 3.1.4.2 on page 41).
4. Optionally, connect the Mediant 2000 RS-232 port to your PC (refer to Section 3.1.4.3 on page 41).
5. Connect the power supply (refer to Section 3.1.4.4 on page 42).

3.1.4.1 Connecting the E1/T1 Trunk Interfaces

Connect the E1/T1 Trunk Interfaces using **either** Telco or RJ-48 connectors:

➢ With 50-pin Telco connectors (16-trunk device), take these 3 steps:

1. Attach the Trunk cable (of at least 26 AWG UTP) with a 50-pin male Telco connector to the 50-pin female Telco connector labeled 'Trunks 1→8' on the Rear Transition Module (RTM).
2. Connect the other end of the Trunk cable to the PBX/PSTN switch.
3. Repeat steps 1 and 2 for the other Trunk cable but this time connect it to the connector labeled 'Trunks 9→16'.

The 50-pin male Telco cable connector must be wired according to the pinout in Table 3-3 below, and to mate with the female connector illustrated in Figure 3-5.

Figure 3-5: 50-pin Female Telco Board-Mounted Connector

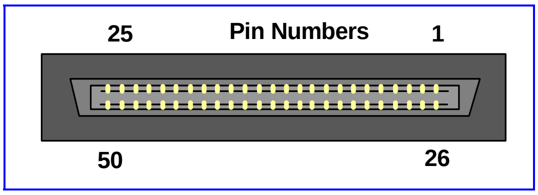


Table 3-3: E1/T1 Connections on each 50-pin Telco Connector

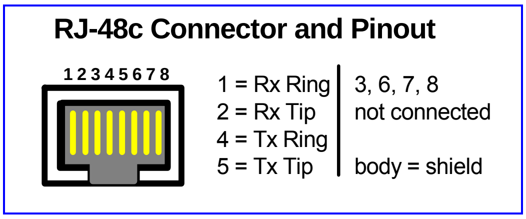
E1/T1 Number		Tx Pins (Tip/Ring)	Rx Pins (Tip/Ring)
1 to 8	9 to 16		
1	9	27/2	26/1
2	10	29/4	28/3
3	11	31/6	30/5
4	12	33/8	32/7
5	13	35/10	34/9
6	14	37/12	36/11
7	15	39/14	38/13
8	16	41/16	40/15

➢ With RJ-48c Connectors, take these 2 steps:

1. Connect the E1/T1 trunk cables to the ports labeled 'Trunks 1 to 8' (in the case of the 8-trunk device) on the RTM.
2. Connect the other ends of the Trunk cables to the PBX/PSTN switch.

RJ-48c trunk connectors are wired according to Figure 3-6 below.

Figure 3-6: Pinout of RJ-48c Trunk Connectors

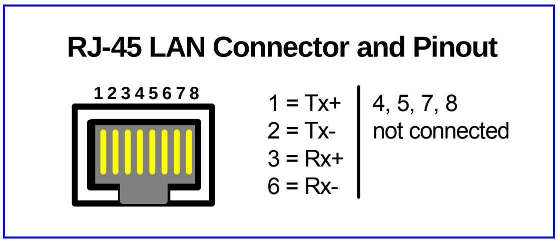


3.1.4.2 Installing the Ethernet Connection

Connect a standard Category 5 network cable to the Ethernet RJ-45 port (and the other as optional redundancy/backup). Connect the other end of the Category 5 network cables to your IP network. The Ethernet connectors (labeled Ethernet 1 and Ethernet 2) are wired according to Figure 3-7.

When assigning an IP address to the gateway using HTTP (under Step 1 in Section 4.2.1), you may be required to disconnect this cable and re-cable it differently.

Figure 3-7: Pinout of RJ-45 Connectors



Note: For redundant operation it is recommended to connect each of the Ethernet connectors to a different switch.

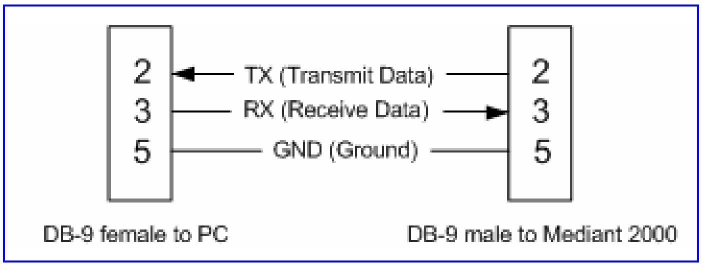
3.1.4.3 Connecting the RS-232 Port to Your PC

Using a standard RS-232 straight cable (not a cross-over cable) with DB-9 connectors, connect the RS-232 port to either COM1 or COM2 RS-232 communication port on your PC. The required connector pinout and gender are shown below in Figure 3-8.

Note that the RS-232 port is available only on the 1, 2 and 4-span configuration.

For information on establishing a serial communications link with the gateway, refer to Section 11.2 on page 248.

Figure 3-8: RS-232 Cable Wiring



3.1.4.4 Connecting the Power Supply

Connect the Mediant 2000 to the power supply using one of the following methods:

3.1.4.4.1 Connecting the AC Power Supply

➢ **When using a single AC power cable, take this step:**

- Attach one end of the supplied 100/240 VAC power cable to the rear AC socket and connect the other end to the correct earthed AC power supply.

➢ **When using a dual AC power cable, take this step:**

- Attach one end of the supplied 100/240 VAC power cables to the rear AC sockets and connect the other end to a separate earthed mains circuits (for power source redundancy).

Note: For the dual AC power supply note the following:

- The LED on the left side of the chassis is only connected when the dual AC is used. It is not relevant to the single AC power connection.
- If only a single socket is connected to the AC power, (while the other plug is left unconnected) the chassis' LED (on the left side) is lit Red, indicating that one of the dual power inlets is disconnected.
- When both the AC power cables are connected, one of the plugs can be disconnected under power without affecting operation, in which case the chassis' left LED is lit Red.
- UPS can be connected to either (or both) of the AC connections.
- The dual AC connections operate in a 1 + 1 configuration and provide load-sharing redundancy.
- Each of the dual power cables can be connected to different AC power phases.

3.1.4.4.2 Connecting the DC Power Supply

➢ **To connect the Mediant 2000 to a DC power supply use one of these two options:**

- DC Terminal block with a screw connection type.
- DC Terminal block with a crimp connection type.

➢ **When using a DC terminal block screw connector:**

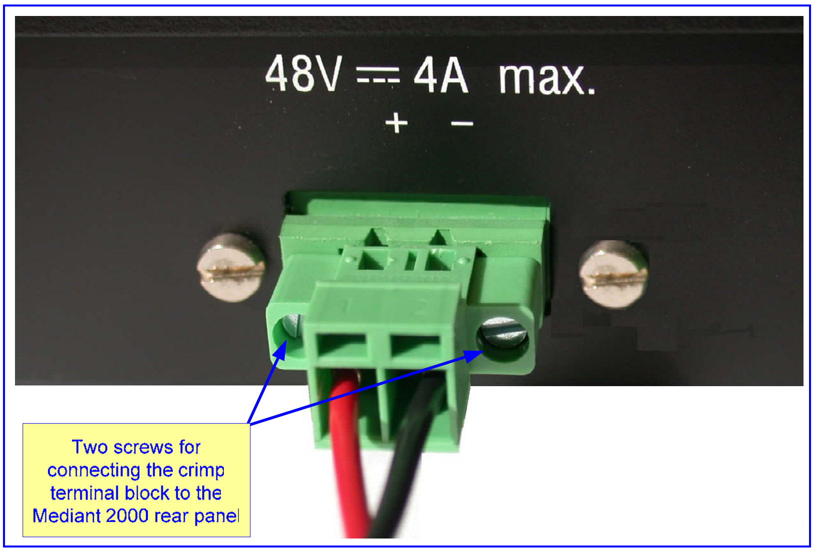
1. Create a DC cable by inserting two 14-16 AWG insulated wires into the supplied adaptor (refer to Figure 3-9) and fasten the two screws, each one located directly above each wire.
2. Connect the two insulated wires to the correct DC power supply. Ensure that the connections to the DC power supply maintain the correct polarity.
3. Insert the terminal block into the DC inlet located on the Mediant 2000.

Figure 3-9: DC Terminal Block Screw Connector

➢ When using a DC terminal block crimp connector:

1. Remove the DC adaptor (screw connection type) that is attached to the Mediant 2000 rear panel.
2. Connect the two insulated wires to the correct DC power supply. Ensure that the connections to the DC power supply maintain the correct polarity (refer to Figure 3-10).
3. Insert the terminal block into the DC inlet located on the Mediant 2000.

Figure 3-10: DC Terminal Block Crimp Connector



3.2 Installing the TP-1610

Electrical Earthing

Prior to installation of any board in a chassis, always correctly connect the chassis to a safety ground according to the laws and regulations of the country in which the installation is performed.

Electrical Component Sensitivity

Electronic components on printed circuit boards are extremely sensitive to static electricity. Normal amounts of static electricity generated by clothing can damage electronic equipment. To reduce the risk of damage due to electrostatic discharge when installing or servicing electronic equipment, it is recommended that anti-static earthing straps and mats be used.

➢ **To install the TP-1610, take these 4 steps:**

1. Unpack the TP-1610 (refer to Section 3.2.1 below).
2. Check the package contents (refer to Section 3.2.2 below).
3. Install the TP-1610 in your PC (refer to Section 3.2.3).
4. Cable the TP-1610 (refer to Section 3.2.4 on page 45).

When you have completed the above relevant sections you are then ready to start configuring the gateway (Chapter 4 on page 49).

3.2.1 Unpacking

➢ **To unpack the TP-1610, take these 6 steps:**

1. Open the carton and remove packing materials.
2. Remove the TP-1610 board from the carton.
3. Check that there is no equipment damage.
4. Check, retain and process any documents.
5. Notify AudioCodes or your local supplier of any damage or discrepancies.
6. Retain any diskettes or CDs.

3.2.2 Package Contents

Ensure that in addition to the TP-1610, the package contains:

- Optionally, an RTM module.
- CD (software and documentation).
- This User's Manual.
- The Mediant & TP Series SIP Digital Gateways Release Notes

3.2.3 Installing the TP-1610

The TP-1610 cPCI board is hot-swappable and can therefore be removed from a slot (and inserted into a slot) while the chassis is under power. It is recommended though that you power down the chassis and read the notes below before replacing the components.

Notes:

- Before removing or inserting boards from / to the chassis, attach a wrist strap for electrostatic discharge (ESD) and connect it to the rack frame using an alligator clip.
- Do not set components down without protecting them with a static bag.

3.2.3.1 Inserting Boards

➢ To insert the TP-1610 board into the chassis, take these 3 steps:

1. Choose an available slot in a compactPCI™ chassis and gently insert the TP-1610 board into it; as the TP-1610 board is inserted, the black plastic handles, at both ends of the board's front panel, must engage with the chassis. When the TP-1610 board is firmly mounted into the correct position inside the chassis, the red plastic latches within each handle self-lock (this also ensures that the TP-1610 board is properly earthed via the chassis) and the blue hot-swap LED is lit.
2. Wait for the blue hot-swap LED to turn off, indicating that the board has been inserted correctly and the power supply is functioning correctly.
3. Fasten the screws on the front panel of the board to secure the board to the chassis.

➢ To insert the TP-1610 RTM into the chassis, take these 2 steps:

1. Choose an available slot in a compactPCI™ chassis and gently insert the TP-1610 RTM into it; as the TP-1610 RTM is inserted, the black plastic handles, at both ends of the board's panel, must engage with the chassis. When the TP-1610 RTM is firmly mounted into the correct position inside the chassis, the red plastic latches within each handle self-lock (this also ensures that the TP-1610 board is properly earthed via the chassis).
2. Fasten the screws on the front panel of the board to secure the board to the chassis.

3.2.3.2 Removing Boards

➢ To remove the TP-1610 board from the chassis, take these 3 steps:

1. Unfasten the screws on the plate of the board.
2. Press the red ejector buttons on the two black ejector/injector latches on both ends and wait for the hot-swap blue LED to light, indicating that the board can be removed.
3. Pull on the two ejector/injector latches and ease out the board from the slot.

➢ To remove the TP-1610 RTM from the chassis, take these 4 steps:

1. Remove the cables attached to the RTM.
2. Unfasten the screws on the brackets at both ends of the panel that secure the RTM to the chassis.
3. Press the red ejector buttons on the two black ejector/injector latches on both ends.
4. Grasp the panel and ease the RTM board out of the slot.

3.2.4 Cabling the TP-1610

➢ To cable the TP-1610, take these 3 steps:

1. Connect the E1/T1 trunk interfaces (refer to Section 3.1.4.1 on page 40).
2. Install the Ethernet connection (refer to Section 3.1.4.2 on page 41).
3. Optionally, connect the TP-1610 RS-232 port to your PC (refer to Section 3.1.4.3 on page 41).

3.3 Installing the TP-260

Electrical Earthing

Prior to installation of any board in a chassis, always correctly connect the chassis to a safety ground according to the laws and regulations of the country in which the installation is performed.

Electrical Component Sensitivity

Electronic components on printed circuit boards are extremely sensitive to static electricity. Normal amounts of static electricity generated by clothing can damage electronic equipment. To reduce the risk of damage due to electrostatic discharge when installing or servicing electronic equipment, it is recommended that anti-static earthing straps and mats be used.

➢ To install the TP-260, take these 4 steps:

1. Unpack the TP-260 (refer to Section 3.3.1 below).
2. Check the package contents (refer to Section 3.3.2 below).
3. Install the TP-260 in your PC (refer to Section 3.3.3 on page 47).
4. Cable the TP-260 (refer to Section 3.3.4 on page 47).

When you have completed the above relevant sections you are then ready to start configuring the gateway (Chapter 4 on page 49).

3.3.1 Unpacking

➢ To unpack the TP-260, take these 6 steps:

1. Open the carton and remove packing materials.
2. Remove the TP-260 board from the carton.
3. Check that there is no equipment damage.
4. Check, retain and process any documents.
5. Notify AudioCodes or your local supplier of any damage or discrepancies.
6. Retain any diskettes or CDs.

3.3.2 Package Contents

Ensure that in addition to the TP-260, the package contains:

- For the 8 span version only, four E1/T1 cable splitters (shown in Figure 3-11).
- CD (software and documentation).
- This User's Manual.
- The Mediant & TP Series SIP Digital Gateways Release Notes

3.3.3 Installing the TP-260

➢ To install the TP-260 board, take these 8 steps:

1. End all applications running in the PC.
2. Shut down the PC, turn off the power, and remove the PC's cover.
3. Choose an available PCI slot and remove its blank rear bracket.
4. Insert the TP-260 board into the chosen PCI slot. Ensure that the front panel (bracket) of the TP-260 board fits correctly into the opening in the rear panel of the PC's chassis. Also, check that the edge of the PCI retainer bracket fits correctly into the PC’s PCI slot.
5. Secure the front panel of the TP-260 board into the chassis frame with a standard screw. This also ensures chassis ground to the TP-260 board.
6. Replace and secure the PC's cover.
7. Power up the PC.
8. When using Windows™ operating systems, the PC prompts that new hardware has been found. The driver for the TP-260 is found in the supplied software package (*260_UNSeries.inf*).
   Note that since the TP-260 PCI gateway operates independently and relies on the host’s PCI only for its power, the driver is only used to prevent the Found new Hardware Wizard to reappear each time the host PC restarts.

3.3.4 Cabling the TP-260

➢ To cable the TP-260, take these 2 steps:

1. Connect the TP-260 E1/T1 interfaces to your E1/T1 trunks by using the four supplied TP-260 E1/T1 cable splitters (shown in Figure 3-11). Connect a splitter to each of the four RJ-48 connectors labeled Trunks 1/5, 2/6, 3/7 and 4/8 on the TP-260 front panel. Each splitter distributes each RJ-48 connector into two separate connectors (wired according to Figure 2-7): the first connector (labeled 1/4) on each splitter supports each of the first four trunks, the second connector (labeled 5/8) on each splitter supports each of the last four trunks.

> **Note:** The TP-260 E1/T1 cable splitter is part of the 8-span TP-260 product and is PSTN certified. When using a *non*-AudioCodes E1/T1 cable splitter, AudioCodes cannot guarantee compliance with PSTN homologations.

Figure 3-11: TP-260 E1/T1 Cable Splitter

2. Connect the TP-260 Ethernet connection, located on the front panel, directly to the network using a standard RJ-45 Ethernet cable. The Ethernet connector is wired according to Figure 2-6.
   Note that when assigning an IP address to the TP-260 using HTTP (under Step 1 in Section 4.2.1), you may be required to disconnect this cable and re-cable it differently.

Reader's Notes

4 Getting Started

The Mediant 2000 is composed of one or two identical media gateway modules. These media gateways are fully independent, each gateway having its own MAC and IP addresses (Table 4-1 shows the default IP addresses of the Mediant 2000).

Before you begin configuring each gateway, change its default IP address to correspond with your network environment (refer to Section 4.2) and learn about the configuration methods available on the Mediant 2000 (refer to Section 4.1 below).

For information on quickly setting up the gateway with basic parameters using a standard Web browser, refer to Section 4.3 on page 53.

Table 4-1: Default Networking Parameters

Product Version	Default Value
Mediant 2000 and TP-1610 with a single module (up to 8 trunks) configuration, TP-260	10.1.10.10
Mediant 2000 and TP-1610 with a double module (16 trunks) configuration	10.1.10.10 (trunks 1-8) and 10.1.10.11 (trunks 9-16)
Default subnet mask is 255.255.0.0, default gateway IP address is 0.0.0.0	

4.1 Configuration Concepts

Users can utilize the gateway in a wide variety of applications, enabled by its parameters and configuration files (e.g., Call Progress Tones (CPT)). The parameters can be configured and configuration files can be loaded using:

- A standard Web Browser (described and explained in Section 5 on page 55).
- A configuration file referred to as the *ini* file. For information on how to use the *ini* file, refer to Section 6 on page 127.
- An SNMP browser software (refer to Section 15 on page 307).
- AudioCodes' Element Management System (EMS) (refer to Section 15.10 on page 326 and to AudioCodes' EMS User's Manual or EMS Product Description). (Doesn't apply to the TP-260).

To upgrade the gateway (load new software or configuration files onto the gateway) use the Software Upgrade wizard, available through the Web Interface (refer to Section 5.8.1 on page 115), or alternatively use the BootP/TFTP configuration utility (refer to Section 7.3.1 on page 205).

For information on the configuration files, refer to Section 16 on page 329.

4.2 Assigning an IP Address to the Gateway

To assign an IP address to each of the Mediant 2000 modules use one of the following methods:

- HTTP using a Web browser (refer to Section 4.2.1 below).
- BootP (refer to Section 4.2.2 on page 51).
- The embedded Command Line Interface (CLI) accessed via RS-232 (if supported) or Telnet (refer to Section 4.2.3 on page 51).
- Dynamic Host Control Protocol (DHCP) (refer to Section 7.2 on page 204).

Use the 'Reset' button at any time to restore the gateway networking parameters to their factory default values (refer to Section 11.1 on page 247).

4.2.1 Assigning an IP Address Using HTTP

➢ To assign an IP address using HTTP, take these 9 steps:

1. Disconnect the gateway from the network and reconnect it to your PC using one of the following two methods:
   - Use a standard Ethernet cable to connect the network interface on your PC to a port on a network hub / switch. Use a second standard Ethernet cable to connect the gateway to another port on the same network hub / switch.
   - Use an Ethernet cross-over cable to directly connect the network interface on your PC to the gateway.
2. Change your PC's IP address and subnet mask to correspond with the gateway's factory default IP address and subnet mask, shown in Table 4-1. For details on changing the IP address and subnet mask of your PC, refer to Windows™ Online Help (Start>Help).
3. Access the gateway's Embedded Web Server (refer to Section 5.3 on page 58).
4. In the 'Quick Setup' screen (shown in Figure 4-1), set the gateway 'IP Address', 'Subnet Mask' and 'Default Gateway IP Address' fields under 'IP Configuration' to correspond with your network IP settings. If your network doesn't feature a default gateway, enter a dummy value in the 'Default Gateway IP Address' field.
5. Click the **Reset** button and click **OK** at the prompt; the gateway applies the changes and restarts.

Tip: Record and retain the IP address and subnet mask you assign the gateway. Do the same when defining new username or password. If the Embedded Web Server is unavailable (for example, if you've lost your username and password), use the BootP/TFTP (Trivial File Transfer Protocol) configuration utility to access the device, 'reflash' the load and reset the password (refer to Appendix D on page 353 for detailed information on using a BootP/TFTP configuration utility to access the device).

6. When implementing a Mediant 2000 with two modules, repeat steps 3 to 5 for the second module; otherwise, skip to Step 7.
7. Disconnect your PC from the gateway or hub / switch (depending on the connection method you used in Step 1).
8. Reconnect the gateway and your PC (if necessary) to the network.
9. Restore your PC's IP address and subnet mask to what they originally were. If necessary, restart your PC and re-access the gateway via the Embedded Web Server with its new assigned IP address.

4.2.2 Assigning an IP Address Using BootP

Note: BootP procedure can also be performed using any standard compatible BootP server.

Tip: You can also use BootP to load the auxiliary files to the Mediant 2000 (refer to Section 6.18 on page 201).

➢ **To assign an IP address using BootP, take these 4 steps:**

1. Open the BootP application (supplied with the Mediant 2000 software package).
2. Add client configuration for the gateway that you want to initialize, refer to Section D.11.1 on page 360.
3. Use the reset button to *physically* reset the gateway causing it to use BootP; the Mediant 2000 changes its network parameters to the values provided by the BootP.
4. Repeat steps 2 and 3 for the Mediant 2000 second module (if used).

4.2.3 Assigning an IP Address Using the CLI

First access the CLI using a standard Telnet application or using a serial communication software (e.g., HyperTerminal™) connected to the RS-232 port (if supported) (refer to Section 4.2.3.1 below). Then assign the Mediant 2000 an IP address (refer to Section 4.2.3.2 below).

4.2.3.1 Accessing the CLI

➢ **To access the CLI via the Embedded Telnet Server, take these 3 steps:**

1. Enable the Embedded Telnet Server:
   - Access the Mediant 2000 Embedded Web Server (refer to Section 5.3 on page 58).
   - Set the parameter 'Embedded Telnet Server' (under **Advanced Configuration** > **Network Settings** > **Application Settings**) to 'Enable (Unsecured)' or 'Enable Secured (SSL)'.
   - Click the **Maintenance** button on the main menu bar; the 'Maintenance Actions' screen is displayed.
   - From the 'Burn to FLASH' drop-down list, select 'Yes', and then click the **Reset** button; the IPmedia 2000 is shut down and re-activated. A message about the waiting period is displayed. The screen is refreshed.
2. Use a standard Telnet application to connect to the Mediant 2000 Embedded Telnet Server. Note that if the Telnet server is set to SSL mode, a special Telnet client is required on your PC to connect to the Telnet interface over a secured connection.
3. Login using the username ('Admin') and password ('Admin').

➢ **To access the CLI via the RS-232 port, take these 2 steps:**

1. Connect the RS-232 port to your PC (For the Mediant 2000, refer to Section 3.1.4.3 on page 41.
2. Use serial communication software (e.g., HyperTerminal™) to connect to the Mediant 2000.
   Set your serial communication software to the following communications port settings:
   - Baud Rate: 115,200 bps
   - Data bits: 8
   - Parity: None
   - Stop bits: 1
   - Flow control: None

The CLI prompt becomes available.

4.2.3.2 Assign an IP Address

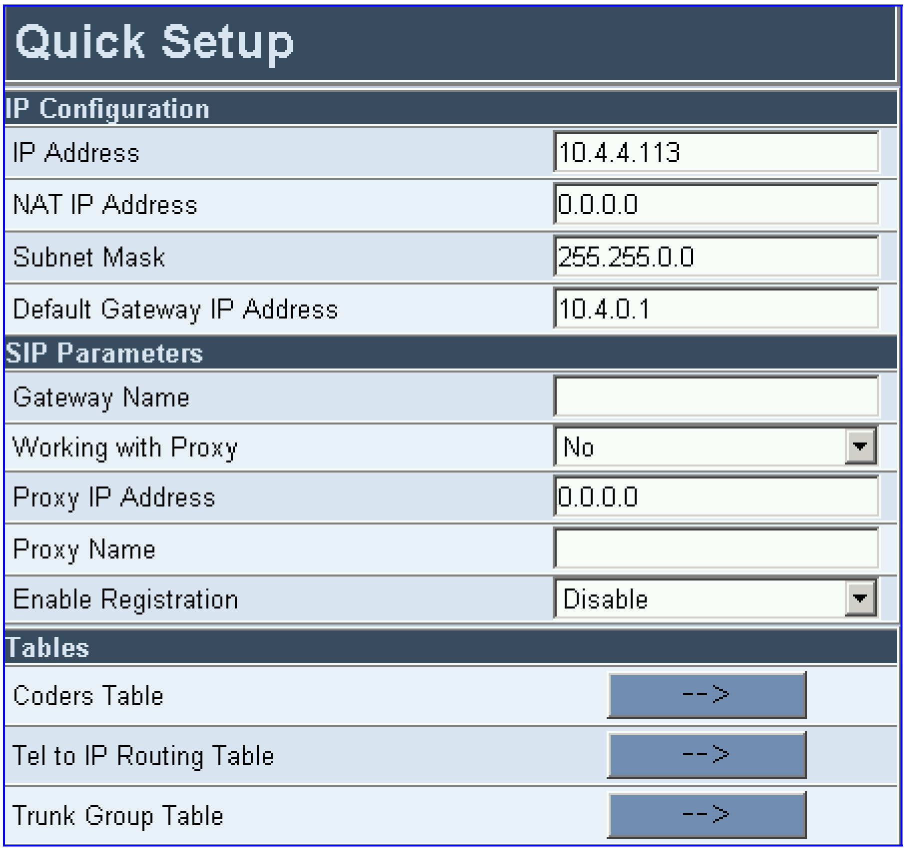
➢ **To assign an IP address via the CLI, take these 4 steps:**

1. At the prompt type **`conf`**, and then press Enter; the configuration folder is accessed.
2. To check the current network parameters, at the prompt, type **`GCP IP`**, and then press Enter; the current network settings are displayed.
3. Change the network settings by typing:
   `SCP IP` `[ip_address] [subnet_mask] [default_gateway]`
   (e.g., 'SCP IP 10.13.77.7 255.255.0.0 10.13.0.1'); the new settings take effect on-the-fly. Connectivity is active at the new IP address.
   Note: This command requires you to enter all three network parameters (each separated by a space).
4. To save the configuration, at the prompt, type **`SAR`**, and then press Enter; the Mediant 2000 restarts with the new network settings.

4.3 Configuring the Gateway's *Basic* Parameters

To configure the gateway's *basic* parameters use the Embedded Web Server's 'Quick Setup' screen (shown in Figure 4-1 below). Refer to Section 5.3 on page 58 for information on accessing the 'Quick Setup' screen.

Figure 4-1: Quick Setup Screen



➢ To configure basic SIP parameters, take these 10 steps:

1. If the gateway is connected to a router with NAT (Network Address Translation) enabled, perform the following procedure. If it isn't, leave the 'NAT IP Address' field undefined.
   - Determine the 'public' IP address assigned to the router (by using, for instance, router Web management). Enter this public IP address in the 'NAT IP Address' field.
   - Enable the DMZ (Demilitarized Zone) configuration on the residential router for the LAN port where the gateway gateway is connected. This enables unknown packets to be routed to the DMZ port.
2. Under 'SIP Parameters', enter the gateway's domain name in the field 'Gateway Name'. If the field is not specified, the gateway's IP address is used instead (default).
3. When working with a Proxy server, set 'Working with Proxy' field to 'Yes' and enter the IP address of the primary Proxy server in the field 'Proxy IP address'. When no Proxy is used, the internal routing table is used to route the calls.
4. Enter the Proxy name in the field 'Proxy Name'. If Proxy name is used, it replaces the Proxy IP address in all SIP messages. This means that messages is still sent to the physical Proxy IP address but the SIP URI contains the Proxy name instead.

5. Configure 'Enable Registration' to 'Yes' or 'No':
'No' = the gateway does not register to a Proxy server/Registrar (default).
'Yes' = the gateway registers to a Proxy server/Registrar at power up and every 'Registration Time' seconds. For detailed information on the parameter 'Registration Time', refer to Table 6-7 on page 150.
6. To program the Coders Table, click the arrow button next to 'Coders Table'. For information on how to configure the Coders Table, refer to Section 5.5.1.1 on page 63.
7. To program the Tel to IP Routing Table, click the arrow button next to 'Tel to IP Routing Table'. For information on how to configure the Tel to IP Routing Table, refer to Section 5.5.5.1 on page 70.
8. To program the E1/T1 B-channels, click the arrow button next to 'Trunk Group Table'. For information on how to configure the Trunk Group Table, refer to Section 5.5.7 on page 82.
9. Click the Reset button and click OK at the prompt; the gateway applies the changes and restarts.
10. After the gateway was reset, access the Advanced Configuration>Trunk Settings page, and select the gateway's E1/T1 protocol type and Framing method that best suits your system requirements. Note that for E1 spans, the framing method must always be set to 'Extended Super Frame'. For information on how to configure the Trunk Settings, refer to Section 5.6.3 on page 88.

You are now ready to start using the gateway. To prevent unauthorized access to the gateway, it is recommended that you change the username and password that are used to access the Web Interface. Refer to Section 5.6.8.1 on page 98 for details on how to change the username and password.

Tip: Once the gateway is configured correctly back up your settings by making a copy of the VoIP gateway configuration (*ini* file) and store it in a directory on your PC. This saved file can be used to restore configuration settings at a future time. For information on backing up and restoring the gateway's configuration, refer to Section 5.6.6 on page 96.

5 Web Management

The Embedded Web Server is used both for gateway configuration, including loading of configuration files, and for run-time monitoring. The Embedded Web Server can be accessed from a standard Web browser, such as Microsoft™ Internet Explorer, Netscape™ Navigator, etc. Specifically, users can employ this facility to set up the gateway configuration parameters. Users also have the option to remotely reset the gateway and to permanently apply the new set of parameters.

5.1 Computer Requirements

To use the Embedded Web Server, the following is required:

- A computer capable of running your Web browser.
- A network connection to the VoIP gateway.
- One of the following compatible Web browsers:
    - Microsoft™ Internet Explorer™ (version 6.0 and higher).
    - Netscape™ Navigator™ (version 7.2 and higher).

Note: The browser must be Java-script enabled. If java-script is disabled, access to the Embedded Web Server is denied.

5.2 Protection and Security Mechanisms

Access to the Embedded Web Server is controlled by the following protection and security mechanisms:

- User accounts (refer to Section 5.2.1 below).
- Read-only mode (refer to Section 5.2.2 below).
- Disabling access (refer to Section 5.2.3 below).
- Secured HTTP connection (HTTPS) (refer to Section 13.2.2 on page 290).
- Limiting access to a predefined list of IP addresses (refer to Section 5.6.8.2 on page 100).
- Managed access using a RADIUS server (refer to Section 13.3 on page 294).

5.2.1 User Accounts

To prevent unauthorized access to the Embedded Web Server, two user accounts are available, a primary and secondary. Each account is composed of three attributes: username, password and access level. The username and password enable access to the Embedded Web Server itself; the access level determines the extent of the access (i.e., availability of screens and read / write privileges). Note that additional accounts can be defined using a RADIUS server (refer to Section 13.3 on page 294).

Table 5-1 lists the available access levels and their privileges.

Table 5-1: Available Access Levels and their Privileges

Access Level	Numeric Representation*	Privileges
Security Administrator	200	Read / write privileges for all screens
Administrator	100	Read-only privilege for security-related screens and read / write privileges for the others
User Monitor	50	No access to security-related and file-loading screens and read-only access to the others
No Access	0	No access to any screen

* The numeric representation of the access level is used only to define accounts in a RADIUS server (the access level ranges from 1 to 255).

The access level mechanism operation is as follows (for both Web and RADIUS accounts):

Each Web screen features two (hard-coded) minimum access levels, read and write. The read access level determines whether the screen can be viewed. The write access level determines whether the information in the screen can be modified.

When a user tries to access a specific Web screen, his access level is compared with the access levels of the screen:

- If the access level of the user is less than the screen's read access level, the screen cannot be viewed.
- If the access level of the user is equal to or greater than the screen's read access level but less than the write access level, the screen is read only.
- If the access level of the user is equal to or greater than the screen's write access level, the screen can be modified.

The default attributes for the two accounts are shown in Table 5-2 below:

Table 5-2: Default Attributes for the Accounts

Account / Attribute	Username (Case-Sensitive)	Password (Case-Sensitive)	Access Level
Primary Account	Admin	Admin	Security Administrator*
Secondary Account	User	User	User Monitor

* The access level of the primary account cannot be changed; all other account-attributes can be modified.

The first time a browser request is made, users are requested to provide their account's username and password to obtain access. If the Embedded Web Server is left idle for more than five minutes, the session expires and the user is required to re-enter his username and password.

Tip: To access the Embedded Web Server with a different account, click the **Log Off** button and re-access with a new username and password.

For details on changing the account attributes, refer to Section 5.6.8.1 on page 98. Note that the password and username can be a maximum of 19 case-sensitive characters.

To reset the username and password of both accounts to their defaults, set the *ini* file parameter 'ResetWebPassword' to 1.

5.2.2 Limiting the Embedded Web Server to Read-Only Mode

Users can limit access to the Embedded Web Server to read-only mode by changing the *ini* file parameter 'DisableWebConfig' to 1. In this mode all Web screens, regardless of the access level used, are read-only and cannot be modified. In addition, the following screens cannot be accessed: 'Quick Setup', 'Web User Accounts', 'Maintenance Actions', and all of the file-loading screens.

Notes:

- Read-only policy can also be applied to selected users by setting the access level of the secondary account to 'User Monitor' (DisableWebConfig = 0) and distributing the primary and secondary accounts to users according to the organization's security policy.
- When DisableWebConfig is set to 1, read-only privileges are applied to all accounts regardless of their access level.

5.2.3 Disabling the Embedded Web Server

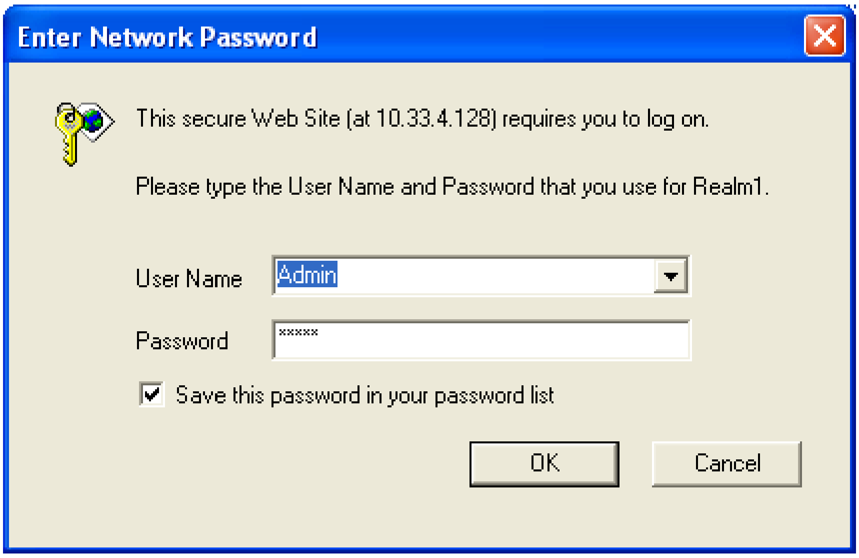
Access to the Embedded Web Server can be disabled by using the *ini* file parameter 'DisableWebTask = 1'. The default is access enabled.

5.3 Accessing the Embedded Web Server

➢ To access the Embedded Web Server, take these 4 steps:

1. Open a standard Web-browsing application such as Microsoft™ Internet Explorer™ or Netscape™ Navigator™.
2. In the Uniform Resource Locator (URL) field, specify the IP address of the gateway (e.g., http://10.1.10.10); the Embedded Web Server's 'Enter Network Password' screen appears, shown in Figure 5-1.

Figure 5-1: Embedded Web Server Login Screen



3. In the 'User Name' and 'Password' fields, enter the username (default: 'Admin') and password (default: 'Admin'). Note that the username and password are case-sensitive.
4. Click the **OK** button; the 'Quick Setup' screen is accessed (shown in Figure 4-1).

5.3.1 Using Internet Explorer to Access the Embedded Web Server

Internet explorer's security settings may block access to the gateway's Web browser if they're configured incorrectly. In this case, the following message is displayed:

Unauthorized
Correct authorization is required for this area. Either your browser does not perform authorization or your authorization has failed. RomPager server.

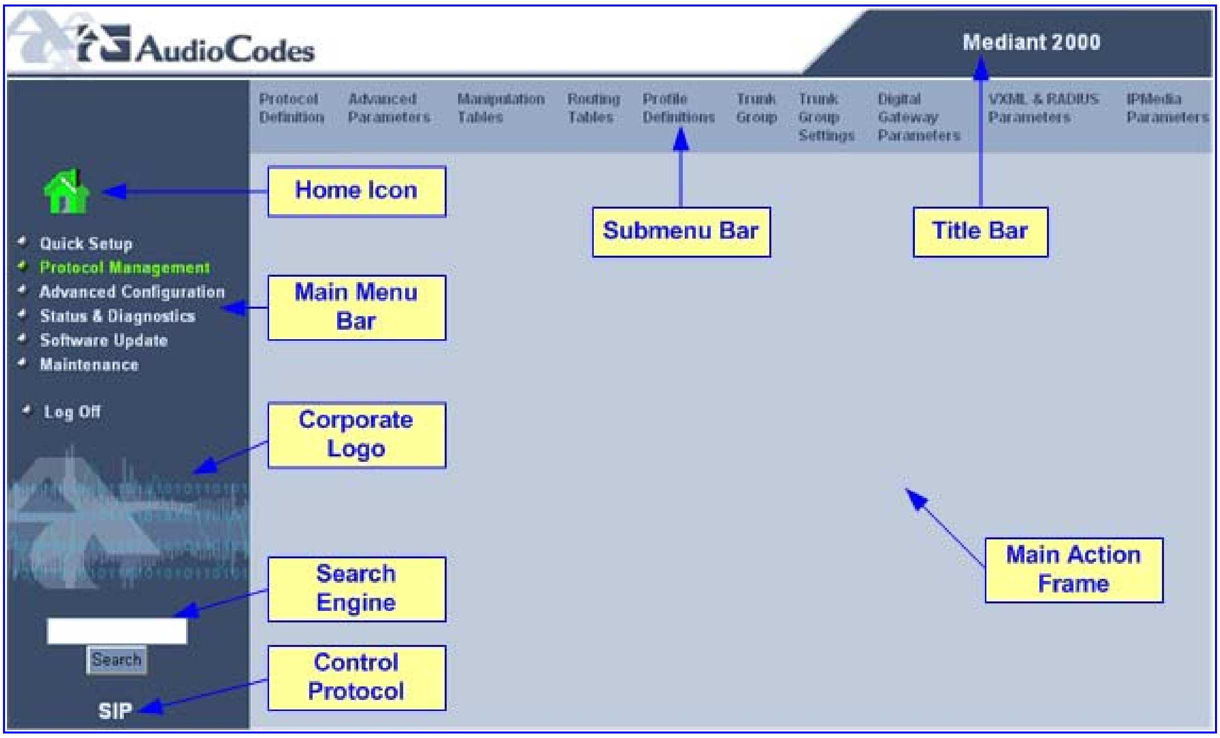
➢ To troubleshoot blocked access to Internet Explorer, take these 3 steps:

1. Delete all cookies from the Temporary Internet files. If this does not clear up the problem, the security settings may need to be altered (continue with Step 2).
2. In Internet Explorer, Tools, Internet Options:
   - Select the Security tab, select Custom Level. Scroll down until the Logon options are displayed and change the setting to: **Prompt for username and password**.
   - Select the Advanced tab, scroll down until the HTTP 1.1 Settings are displayed and verify that the **Use HTTP 1.1** option is checked.
3. Restart the browser.

5.4 Getting Acquainted with the Web Interface

Figure 5-2 shows the general layout of the Web Interface screen.

Figure 5-2: Web Interface



The Web Interface screen features the following components:

- **Title bar:** contains three configurable elements: corporate logo, a background image and the product's name. For information on how to modify these elements, refer to Section 11.6 on page 257.
- **Product name:** the gateway's module name (Module 1 or Module 2).
- **Main menu bar:** always appears on the left of every screen to quickly access parameters, submenus, submenu options, functions and operations.
- **Submenu bar:** appears on the top of screens and contains submenu options.
- **Main action frame:** the main area of the screen in which information is viewed and configured.
- **Home icon:** when clicked it opens the 'Trunk & Channel Status' screen (refer to Section 5.7.2 on page 110).
- **Corporate logo:** AudioCodes' corporate logo. For information on how to remove this logo, refer to Section 11.6 on page 257.
- **Search engine:** for searching *ini* file parameters configurable by using the Embedded Web Server (refer to Section 5.4.3 on page 60).
- **Control Protocol:** the gateway's control protocol.

5.4.1 Main Menu Bar

The main menu bar of the Web Interface is divided into the following 7 menus:

- **Quick Setup:** Use this menu to configure the gateway's basic settings; for the full list of configurable parameters go directly to 'Protocol Management' and 'Advanced Configuration' menus. An example of the Quick Setup configuration is described in Section 4.3 on page 53.
- **Protocol Management:** Use this menu to configure the gateway's control protocol parameters and tables (refer to Section 5.5 on page 63).
- **Advanced Configuration:** Use this menu to set the gateway's advanced configuration parameters (for advanced users only) (refer to Section 5.6 on page 84).
- **Status & Diagnostics:** Use this menu to view and monitor the gateway's channels, Syslog messages, hardware / software product information, and to assess the gateway's statistics and IP connectivity information (refer to Section 5.7 on page 106).
- **Software Update:** Use this menu when you want to load new software or configuration files onto the gateway (refer to Section 5.8 on page 115).
- **Maintenance:** Use this menu to remotely lock/unlock the device (refer to Section 5.9.1 on page 122), save configuration changes to the non-volatile flash memory (refer to Section 5.9.2 on page 124), and reset the gateway (refer to Section 5.9.3 on page 125).

When positioning your curser over a parameter name (or a table) for more than a second, a short description of this parameter is displayed. Note that parameters preceded by an exclamation mark (!) are *not* changeable on-the-fly and require that the device be reset.

5.4.2 Saving Changes

To save changes to the volatile memory (RAM), click the **Submit** button (changes to parameters with on-the-fly capabilities are immediately available; other parameters are updated only after a gateway reset). Parameters that are saved only to the volatile memory revert to their previous settings after hardware reset. When performing a software reset (i.e., via Web or SNMP) you can choose to save the changes to the non-volatile memory. To save changes so they are available after a hardware reset or power fail, you must save the changes to the non-volatile memory (flash memory).

To save the changes to flash, refer to Section 5.9.2 on page 124.

5.4.3 Searching Configuration Parameters

The Embedded Web Server provides a search engine that allows you to search any *ini* file parameter that is configurable by the Web server. The search result provides you a brief description of the parameter as well as a link to the relevant screen in which the parameter is configured in the Web server.

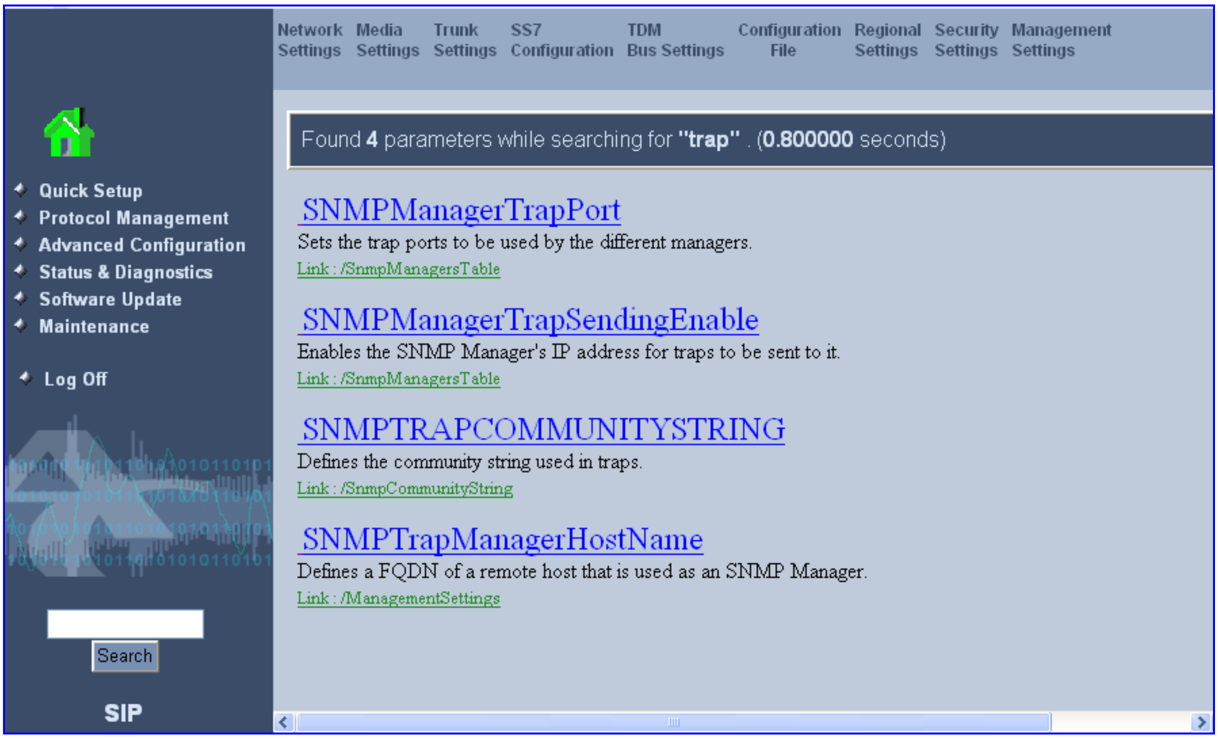
The **Search** button, located near the bottom of the Main menu bar (refer to Figure 5-3) is used to perform parameter searches.

You can search for a specific *ini* parameter (e.g., 'EnableIPSec') or a sub-string of that parameter (e.g., 'sec'). If you search for a sub-string, the Embedded Web Server lists all found parameters that contain the searched sub-string in their parameter names.

➢ To search for an ini file parameter configurable by the Web server, take these 3 steps:

1. In the 'Search' field, enter the name or sub-string of the *ini* parameter for which you want to search.
2. Click **Search**. The 'Searched Result' screen appears, listing all searched parameter results.

Figure 5-3: Searched Result Screen

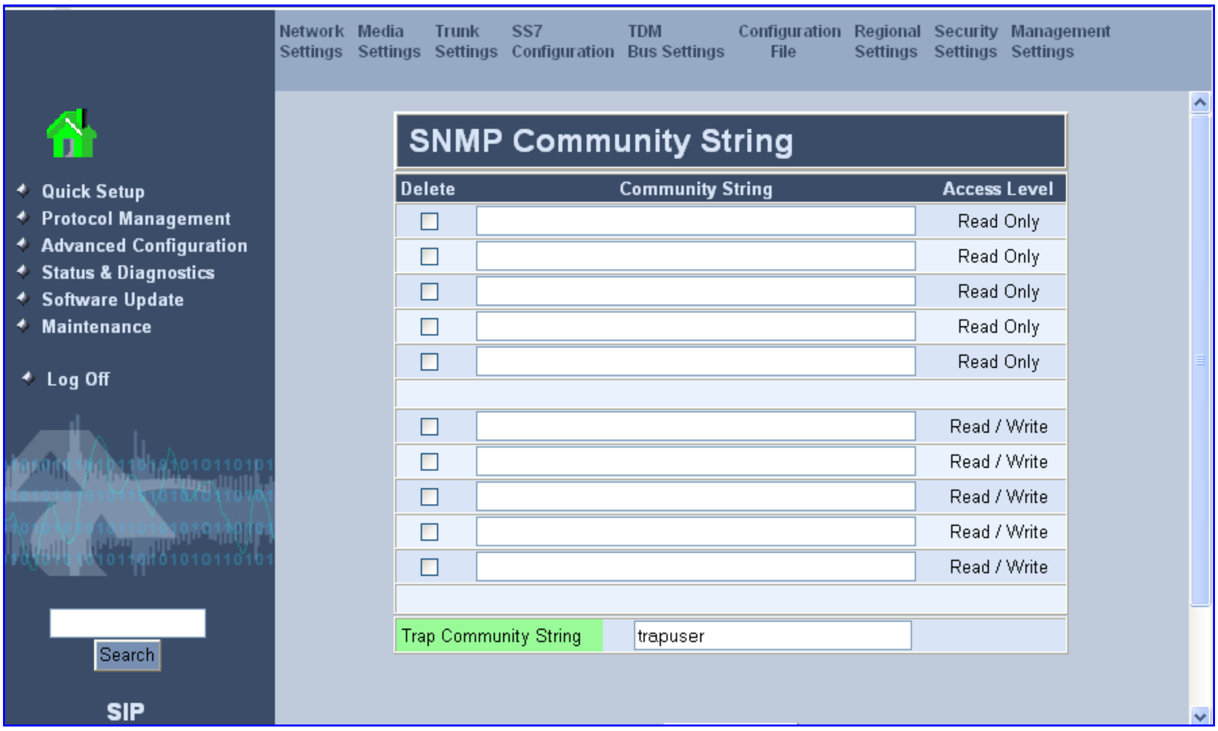


Each searched result displays the following:

- Parameter name (hyperlinked to its location in the Embedded Web Server)
- Brief description of the parameter
- Hyperlink (in green) displaying the URL path to its location in the Embedded Web Server

3. In the Searched Result list, click the required parameter to open the screen in which the parameter appears. In the relevant screen, the serached parameter is highlighted in green for easy viewing.

Figure 5-4: Searched Parameter Highlighted in Screen



Note: If the searched parameter is not located, the "No Matches Found For This String" message is displayed.

5.4.4 Entering Phone Numbers in Various Tables

Phone numbers entered into various tables on the gateway, such as the Tel to IP routing table, must be entered without any formatting characters. For example, if you wish to enter the phone number 555-1212, it must be entered as 5551212 without the hyphen (-). If the hyphen is entered, the entry does not work. The hyphen character is used in number entry only, as part of a range definition. For example, the entry [20-29] means 'all numbers in the range 20 to 29'.

5.5 Protocol Management

Use this menu to configure the gateway's SIP parameters and tables.

5.5.1 Protocol Definition Parameters

Use this submenu to configure the following gateway's specific SIP protocol parameters:

- General Parameters
- Proxy & Registration Parameters
- Coders (refer to Section 5.5.1.1 below)
- DTMF & Dialing Parameters

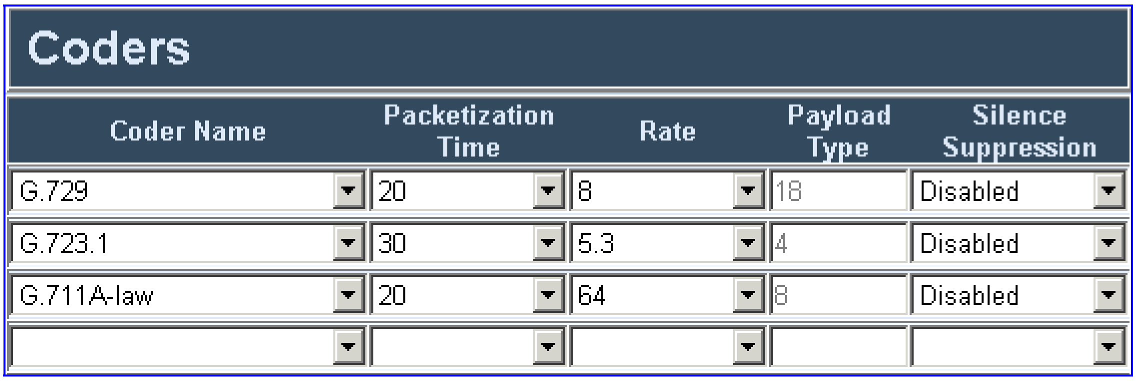
5.5.1.1 Coders

From the Coders screen you can configure the first to fifth preferred coders (and their attributes) for the gateway. The first coder is the highest priority coder and is used by the gateway whenever possible. If the far end gateway cannot use the coder assigned as the first coder, the gateway attempts to use the next coder and so forth.

➢ **To configure the gateway's coders, take these 9 steps:**

1. Open the 'Coders' screen (**Protocol Management** menu > **Protocol Definition** submenu > **Coders** option); the 'Coders' screen is displayed.

Figure 5-5: Coders Screen



2. From the Coder Name drop-down list, select the coder you want to use. For the full list of available coders and their corresponding attributes, refer to the *ini* file parameter 'CoderName' (described in Table 6-7).
   Note: Each coder can appear only once.
3. From the Packetization Time drop-down list, select the packetization time (in msec) for the coder you selected. The packetization time determines how many coder payloads are combined into a single RTP packet.
   Note 1: If not specified, the ptime gets a default value.
   Note 2: The ptime specifies the packetization time the gateway expects to receive. The gateway always uses the ptime requested by the remote side for sending RTP packets.
4. From the Rate drop-down list, select the bit rate (in kbps) for the coder you selected.
5. In the Payload Type field, if the payload type for the coder you selected is dynamic, enter a value from 0 to 120 (payload types of 'well-known' coders cannot be modified). The payload type identifies the format of the RTP payload.
   Note: If not specified, a default is used.

6. From the Silence Suppression drop-down list, enable or disable the silence suppression option for the coder you selected.
   Note: For G.729 it is also possible to select silence suppression without adaptations.
7. Repeat steps 2 to 6 for the second to fifth coders (optional).
8. Click the **Submit** button to save your changes.
9. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

Notes:

- Only the ptime of the first coder in the defined coder list is declared in INVITE / 200 OK SDP, even if multiple coders are defined.
- If the coder G.729 is selected and silence suppression is enabled (for this coder), the gateway includes the string 'annexb=no' in the SDP of the relevant SIP messages. If silence suppression is set to 'Enable w/o Adaptations', 'annexb=yes' is included. An exception to this logic is when the remote gateway is a Cisco device (IsCiscoSCEMode).

5.5.2 Advanced Parameters

Use this submenu to configure the following gateway's advanced control protocol parameters:

- CDR and Debug
- Miscellaneous Parameters
- Supplementary Services

5.5.3 Number Manipulation Tables

The VoIP gateway provides four Number Manipulation tables for incoming and outgoing calls. These tables are used to modify the destination and source telephone numbers so that the calls can be routed correctly.

The Manipulation Tables are:

- Destination Phone Number Manipulation Table for IP→Tel calls
- Destination Phone Number Manipulation Table for Tel→IP call
- Source Phone Number Manipulation Table for IP→Tel calls
- Source Phone Number Manipulation Table for Tel→IP calls

Note: Number manipulation can be performed either before or after a routing decision is made. For example, you can route a call to a specific trunk group according to its original number, and then you can remove/add a prefix to that number before it is routed. To control when number manipulation is done, set the 'RouteModeIP2Tel' and the 'RouteModeTel2IP' parameters. For information on these parameters, refer to Table 6-10 on page 180.

Possible uses for number manipulation can be as follows:

- To strip/add dialing plan digits from/to the number. For example, a user could dial 9 in front of each number to indicate an external line. This number (9) can be removed here before (after) the call is setup.
- Assignment of NPI/TON to IP→Tel calls. The VoIP gateway can use a single global setting for NPI/TON classification or it can use the setting in this table on a call by call basis. Control for this is done using **Protocol Management** > **Protocol Definition** > **Destination/Source Number Encoding Type**.
- Allow / disallow Caller ID information to be sent according to destination / source prefixes.

➢ To configure the Number Manipulation tables, take these 5 steps:

1. Open the Number Manipulation screen you want to configure (**Protocol Management** menu > **Manipulation Tables** submenu); the relevant Manipulation table screen is displayed. Figure 5-6 shows the 'Source Phone Number Manipulation Table for Tel→IP calls'.

Figure 5-6: Source Phone Number Manipulation Table for Tel→IP Calls

	Dest. Prefix	Source Prefix	Num of Stripped Digits	Prefix (Suffix) to Add	Number of Digits to Leave	Presentation
1	03	201	0	972		Allowed
2		1001	4	5(23)		Restricted
3		123451001#	0	(8)	4	Not Configured
4		[30-40]xx	(1)	2		Not Configured
5	[6,7,8]	2001	5	3		Not Configured
6						Not Configured

2. In the 'Table Index' drop-down list, select the range of entries that you want to edit (up to 20 entries can be configured for Source Number Manipulation and 50 entries for Destination Number Manipulation).
3. Configure the Number Manipulation table according to Table 5-3.
4. Click the **Submit** button to save your changes.
5. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

Table 5-3: Number Manipulation Parameters (continues on pages 65 to 66)

Parameter	Description
Destination Prefix	Each entry in the Destination Prefix fields represents a destination telephone number prefix. An asterisk (*) represents any number.
Source Prefix	Each entry in the Source Prefix fields represents a source telephone number prefix. An asterisk (*) represents any number.
Source IP	Each entry in the Source IP fields represents the source IP address of the call (obtained from the Contact header in the INVITE message). This column only applies to the 'Destination Phone Number Manipulation Table for IP to Tel'. **Note:** The source IP address can include the 'x' wildcard to represent single digits. For example: 10.8.8.xx represents all the addresses between 10.8.8.10 to 10.8.8.99. The '*' wildcard represents any number between 0 and 255, e.g., 10.8.8.* represents all addresses between 10.8.8.0 and 10.8.8.255.

Table 5-3: Number Manipulation Parameters (continues on pages 65 to 66)

Parameter	Description
The manipulation rules are applied to any incoming call whose: • Destination number prefix matches the prefix defined in the 'Destination Number' field. • Source number prefix matches the prefix defined in the 'Source Prefix' field. • Source IP address matches the IP address defined in the 'Source IP' field (if applicable). Note that number manipulation can be performed using a combination of each of the above criteria, or using each criterion independently. **Note:** For available notations that represent multiple numbers, refer to Section 5.5.3.1 on page 67.	
Num of stripped digits	• Enter the number of digits that you want to remove from the left of the telephone number prefix. For example, if you enter 3 and the phone number is 5551234, the new phone number is 1234. • Enter the number of digits (in brackets) that you want to remove from the right of the telephone number prefix. **Note:** A combination of the two options is allowed (e.g., 2(3)).
Prefix / Suffix to add	• Prefix - Enter the number / string you want to add to the front of the phone number. For example, if you enter 9 and the phone number is 1234, the new number is 91234. • Suffix - Enter the number / string (in brackets) you want to add to the end of the phone number. For example, if you enter (00) and the phone number is 1234, the new number is 123400. **Note:** You can enter a prefix and a suffix in the same field (e.g., 9(00)).
Number of digits to leave	Enter the number of digits that you want to leave from the right.
Note: The manipulation rules are executed in the following order: **1.** Num of stripped digits **2.** Number of digits to leave **3.** Prefix / suffix to add Figure 5-6 on the previous page exemplifies the use of these manipulation rules in the 'Source Phone Number Manipulation Table for Tel→IP Calls': • When destination number equals 035000 and source number equals 20155, the source number is changed to 97220155. • When source number equals 1001876, it is changed to 587623. • Source number 1234510012001 is changed to 20018. • Source number 3122 is changed to 2312.	
NPI	Select the Number Plan assigned to this entry. You can select Unknown **[0]**, Private **[9]** or E.164 Public **[1]**. The default is Unknown. For a detailed list of the available NPI/TON values, refer to Section 5.5.3.2 on page 67.
TON	Select the Number Type assigned to this entry. • If you selected Unknown as the Number Plan, you can select Unknown **[0]**. • If you selected Private as the Number Plan, you can select Unknown **[0]**, Level 2 Regional **[1]**, Level 1 Regional **[2]**, PISN Specific **[3]** or Level 0 Regional (Local) **[4]**. • If you selected E.164 Public as the Number Plan, you can select Unknown **[0]**, International **[1]**, National **[2]**, Network Specific **[3]**, Subscriber **[4]** or Abbreviated **[6]**. The default is Unknown.
Presentation	Select 'Allowed' to send Caller ID information when a call is made using these destination / source prefixes. Select 'Restricted' if you want to restrict Caller ID information for these prefixes.

5.5.3.1 Dialing Plan Notation

The dialing plan notation applies, in addition to the four Manipulation tables, also to Tel→IP Routing table and to IP→Trunk Group Routing table.

When entering a number in the destination and source 'Prefix' columns, you can create an entry that represents multiple numbers using the following notation:

- [n-m] represents a range of numbers
- [n,m] represents multiple numbers. Note that this notation only supports single digit numbers.
- x represents any single digit
- # (that terminates the number) represents the end of a number
- A single asterisk (*) represents any number

For example:

- [5551200-5551300]# represents all numbers from 5551200 to 5551300
- [2,3,4]xxx# represents four-digit numbers that start with 2, 3 or 4
- 54324 represents any number that starts with 54324
- 54324xx# represents a 7 digit number that starts with 54324
- 123[100-200]# represents all numbers from 123100 to 123200.

The VoIP gateway matches the rules starting at the top of the table. For this reason, enter more specific rules above more generic rules. For example, if you enter 551 in entry 1 and 55 in entry 2, the VoIP gateway applies rule 1 to numbers that starts with 551 and applies rule 2 to numbers that start with 550, 552, 553, 554, 555, 556, 557, 558 and 559. However if you enter 55 in entry 1 and 551 in entry 2, the VoIP gateway applies rule 1 to all numbers that start with 55 including numbers that start with 551.

5.5.3.2 Numbering Plans and Type of Number

Numbers are classified by their Numbering Plan Indication (NPI) and their Type of Number (TON). The gateway supports all NPI/TON classifications used in the standard. The list of ISDN ETSI NPI/TON values is shown as follows:

Table 5-4: NPI/TON Values for ISDN ETSI

NPI	TON	Description
Unknown [0]	Unknown [0]	A valid classification, but one that has no information about the numbering plan.
E.164 Public [1]	Unknown [0]	A public number in E.164 format, but no information on what kind of E.164 number.
	International [1]	A public number in complete international E.164 format. For example: 16135551234
	National [2]	A public number in complete national E.164 format. For example: 6135551234
	Subscriber [4]	A public number in complete E.164 format representing a local subscriber. For example: 5551234
Private [9]	Unknown [0]	A private number, but with no further information about the numbering plan
	Level 2 Regional [1]	
	Level 1 Regional [2]	A private number with a location. For example: 3932200
	PISN Specific [3]	
	Level 0 Regional (local) [4]	A private local extension number. For example: 2200

For NI-2 and DMS-100 ISDN variants the valid combinations of TON and NPI for calling and called numbers are (Plan/Type):

- 0/0 - Unknown/Unknown
- 1/1 - International number in ISDN/Telephony numbering plan
- 1/2 - National number in ISDN/Telephony numbering plan
- 1/4 - Subscriber (local) number in ISDN/Telephony numbering plan
- 9/4 - Subscriber (local) number in Private numbering plan

5.5.4 Mapping NPI/TON to Phone-Context

The Phone-Context table is used to configure the mapping of NPI and TON to the Phone-Context SIP parameter. When a call is received from the ISDN, the NPI and TON are compared against the table and the Phone-Context value is used in the outgoing SIP INVITE message. The same mapping occurs when an INVITE with a Phone-Context attribute is received. The Phone-Context parameter appears in the standard SIP headers where a phone number is used (Request-URI, To, From, Diversion).

➢ To configure the Phone-Context tables, take these 6 steps:

1. Open the 'Phone Context Table' screen (**Protocol Management** menu > **Manipulation Tables** submenu > **Phone Context Table** option); the 'Phone Context Table' screen appears, as shown below.

Figure 5-7: Phone Context Table Screen

2. From the ‘Add Phone Context As Prefix' drop-down list, select 'Enable' to add the received Phone-Context parameter as a prefix to outgoing ISDN SETUP Called and Calling numbers, if necessary.
3. From the 'Phone Context Index' drop-down list, select the index number.

4. Configure the Phone Context table according to Table 5-5.
5. Click the **Submit** button to save your changes.
6. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

Notes:

- Several rows with the same NPI-TON or Phone-Context are allowed. In such a scenario, a Tel-to-IP call uses the first match.
- Phone-Context '+' is a unique case as it doesn't appear in the Request-URI as a Phone-Context parameter. Instead, it's added as a prefix to the phone number. The '+' isn't removed from the phone number in the IP-to-Tel direction.

Table 5-5: Phone-Context Parameters

Parameter	Description
NPI	Select the Number Plan assigned to this entry. You can select the following: 0 = Unknown (default) 1 = E.164 Public 9 = Private For a detailed list of the available NPI/TON values, refer to Section 5.5.3.2 on page 67.
TON	Select the Number Type assigned to this entry. • If you selected Unknown as the NPI, you can select Unknown (0) • If you selected Private as the NPI, you can select Unknown (0), Level 2 Regional (1), Level 1 Regional (2), PSTN Specific (3), or Level 0 Regional (Local) (4). • If you selected E.164 Public as the NPI, you can select Unknown (0), International (1), National (2), Network Specific (3), Subscriber (4), or Abbreviated (6).
Phone Context	The Phone-Context SIP URI parameter.

5.5.5 Configuring the Routing Tables

Use this submenu to configure the gateway's IP→Tel and Tel→IP routing tables and their associated parameters.

5.5.5.1 Tel to IP Routing Table

The Tel to IP Routing Table is used to route incoming Tel calls to IP addresses. This routing table associates a called / calling telephone number's prefixes with a destination IP address or with an FQDN (Fully Qualified Domain Name). When a call is routed through the VoIP gateway (Proxy isn't used), the called and calling numbers are compared to the list of prefixes on the IP Routing Table (up to 50 prefixes can be configured); Calls that match these prefixes are sent to the corresponding IP address. If the number dialed does not match these prefixes, the call is not made.

When using a Proxy server, you do not need to configure the Telephone to IP Routing Table. However, if you want to use fallback routing when communication with Proxy is lost, or to use the 'Filter Calls to IP' and IP Security features, or to obtain different SIP URI host names (per called number), you need to configure the IP Routing Table.

Note that for the Tel to IP Routing table to take precedence over a Proxy for routing calls, set the parameter 'PreferRouteTable' to 1. The gateway checks the 'Destination IP Address' field in the 'Tel to IP Routing' table for a match with the outgoing call. Only if a match is not found, a Proxy is used.

Possible uses for Telephone to IP Routing can be as follows:

- Can fallback to internal routing table if there is no communication with the Proxy.
- Call Restriction – (when Proxy isn't used), reject all outgoing Tel→IP calls that are associated with the destination IP address: 0.0.0.0.
- IP Security – When the IP Security feature is enabled (SecureCallFromIP = 1), the VoIP gateway accepts only those IP→Tel calls with a source IP address identical to one of the IP addresses entered in the Telephone to IP Routing Table.
- Filter Calls to IP – When a Proxy is used, the gateway checks the Tel→IP routing table before a telephone number is routed to the Proxy. If the number is not allowed (number isn't listed or a Call Restriction routing rule was applied), the call is released.
- Always Use Routing Table – When this feature is enabled (AlwaysUseRouteTable = 1), even if a Proxy server is used, the SIP URI host name in the sent INVITE message is obtained from this table. Using this feature, users are able to assign a different SIP URI host name for different called and/or calling numbers.
- Assign Profiles to destination address (also when a Proxy is used).
- Alternative Routing – (When Proxy isn't used) an alternative IP destination for telephone number prefixes is available. To associate an alternative IP address to called telephone number prefix, assign it with an additional entry (with a different IP address), or use an FQDN that resolves to two IP addresses. Call is sent to the alternative destination when one of the following occurs:
  - No ping to the initial destination is available, or when poor Quality of Service (QoS) (delay or packet loss, calculated according to previous calls) is detected, or when a DNS host name is not resolved. For detailed information on Alternative Routing, refer to Section 8.3 on page 210.
  - When a release reason that is defined in the 'Reasons for Alternative Tel to IP Routing' table is received. For detailed information on the 'Reasons for Alternative Routing Tables', refer to Section 5.5.5.4 on page 75.

Alternative routing (using this table) is commonly implemented when there is no response to an INVITE message (after INVITE retransmissions). The gateway then issues an internal 408 ‘No Response’ implicit release reason. If this reason is included in the ‘Reasons for Alternative Routing’ table, the gateway immediately initiates a call to the redundant destination using the next matched entry in the ‘Tel to IP Routing’ table. Note that if a domain name in this table is resolved to two IP addresses, the timeout for INVITE retransmissions can be reduced by using the parameter ‘Number of RTX Before Hotswap’.

Note: If the alternative routing destination is the gateway itself, the call can be configured to be routed back to PSTN. This feature is referred to as ‘PSTN Fallback’, meaning that if sufficient voice quality is not available over the IP network, the call is routed through legacy telephony system (PSTN).

Tip: Tel to IP routing can be performed either before or after applying the number manipulation rules. To control when number manipulation is done, set the RouteModeTel2IP parameter. For information on this parameter, refer to refer to Table 6-10 on page 180.

➢ To configure the Tel to IP Routing table, take these 6 steps:

1. Open the ‘Tel to IP Routing’ screen (**Protocol Management** menu > **Routing Tables** submenu > **Tel to IP Routing** option); the ‘Tel to IP Routing’ screen is displayed (shown in Figure 5-8).
2. In the ‘Tel to IP Routing Mode’ field, select the Tel to IP routing mode (refer to Table 6-10).
3. In the ‘Routing Index' drop-down list, select the range of entries that you want to edit.
4. Configure the Tel to IP Routing table according to Table 5-6.
5. Click the **Submit** button to save your changes.
6. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

Figure 5-8: Tel to IP Routing Table Screen

	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Profile ID	Status
1	10	100	10.33.45.63	1	OK
2	20	*	10.33.45.60	1	QOS Low
3	[3,4,6]	*	10.33.45.64	1	OK
4	54324	[1,2]	Domain.com	1	Dns Error
5	9	*	0.0.0.0	2	n/a
6	8xx#	*	10.13.77.7	1	Ping Error
7	*	*	10.13.77.7	1	OK
8					

Table 5-6: Tel to IP Routing Table

Parameter	Description
Destination Phone Prefix	Each entry in the Destination Phone Prefix fields represents a called telephone number prefix. The prefix can be 1 to 19 digits long. An asterisk (*) represents all numbers.
Source Phone Prefix	Each entry in the Source Phone Prefix fields represents a calling telephone number prefix. The prefix can be 1 to 19 digits long. An asterisk (*) represents all numbers.
Any telephone number whose destination number matches the prefix defined in the 'Destination Phone Prefix' field *and* its source number matches the prefix defined in the adjacent 'Source Phone Prefix' field, is sent to the IP address entered in the 'IP Address' field. Note that Tel to IP routing can be performed according to a combination of source and destination phone prefixes, or using each independently. **Note 1:** An additional entry of the same prefixes can be assigned to enable alternative routing. **Note 2:** For available notations that represent multiple numbers, refer to Section 5.5.3.1 on page 67.	
Destination IP Address	In each of the IP Address fields, enter the IP address (and optionally port number) that is assigned to these prefixes. Domain names, such as domain.com, can be used instead of IP addresses. For example: <IP Address>:<Port> To discard outgoing IP calls, enter 0.0.0.0 in this field. **Note:** When using domain names, you must enter a DNS server IP address, or alternatively define these names in the 'Internal DNS Table'.
Profile ID	Enter the number of the IP profile that is assigned to the destination IP address defined in the 'Destination IP Address' field.
Status	A read only field representing the quality of service of the destination IP address. N/A = Alternative Routing feature is disabled. OK = IP route is available Ping Error = No ping to IP destination, route is not available QoS Low = Bad QoS of IP destination, route is not available DNS Error = No DNS resolution (only when domain name is used instead of an IP address).

5.5.5.2 IP to Trunk Group Routing Table

The IP to Trunk Group Routing Table is used to route incoming IP calls to groups of E1/T1 B-channels called trunk groups. Calls are assigned to trunk groups according to any combination of the following three options (or using each independently):

- Destination phone prefix
- Source phone prefix
- Source IP address

The call is then sent to the VoIP gateway channels assigned to that trunk group. The specific channel, within a trunk group, that is assigned to accept the call is determined according to the trunk group's channel selection mode which is defined in the Trunk Group Settings table (Section 5.5.8 on page 83), or according to the global parameter 'ChannelSelectMode' (refer to Table 6-10 on page 180).

Note: When a release reason that is defined in the 'Reasons for Alternative IP to Tel Routing' table is received for a specific IP→Tel call, an alternative trunk group for that call is available. To associate an alternative trunk group to an incoming IP call, assign it with an additional entry in the 'IP to Trunk Group Routing' table (repeat the same routing rules with a different trunk group ID). For detailed information on the 'Reasons for Alternative Routing Tables', refer to Section 5.5.5.4 on page 75.

To use trunk groups you must also do the following:

- You must assign a trunk group ID to the VoIP gateway E1/T1 B-channels on the Trunk Group Table. For information on how to assign a trunk group ID to a B-channel, refer to Section 5.5.7 on page 82.
- You can configure the Trunk Group Settings table to determine the method in which new calls are assigned to channels within the trunk groups (a different method for each trunk group can be configured). For information on how to enable this option, refer to Section 5.5.8 on page 83. If a Channel Select Mode for a specific trunk group isn't specified, then the global 'Channel Select Mode' parameter (defined in 'General Parameters' screen under 'Advanced Parameters') applies.

➢ To configure the IP to Trunk Group Routing table, take these 6 steps:

1. Open the 'IP to Trunk Group Routing' screen (**Protocol Management** menu > **Routing Tables** submenu > **IP to Trunk Group Routing** option); the 'IP to Trunk Group Routing' table screen is displayed.

Figure 5-9: IP to Trunk Group Routing Table

	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	Trunk Group ID	Profile ID
1	10	*	0	1	2
2	20	101	0	1	2
3					
4					
5	[5010-5020]	*	0	3	1
6	6xx	*	0	3	1
7	71234#	*	0	3	1
8	*	*	0	4	3
9					
10					
11					
12					

2. In the 'IP to Tel Routing Mode' field, select the IP to Tel routing mode (refer to Table 6-10 on page 180).
3. In the 'Routing Index' drop-down list, select the range of entries that you want to edit (up to 24 entries can be configured).
4. Configure the IP to Trunk Group Routing table according to Table 5-7.
5. Click the **Submit** button to save your changes.
6. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

Table 5-7: IP to Trunk Group Routing Table (continues on pages 73 to 74)

Parameter	Description
Destination Phone Prefix	Each entry in the Destination Phone Prefix fields represents a called telephone number prefix. The prefix can be 1 to 49 digits long. An asterisk (*) represents all numbers.
Source Phone Prefix	Each entry in the Source Phone Prefix fields represents a calling telephone number prefix. The prefix can be 1 to 49 digits long. An asterisk (*) represents all numbers.

Table 5-7: IP to Trunk Group Routing Table (continues on pages 73 to 74)

Parameter	Description
Source IP Address	Each entry in the Source IP Address fields represents the source IP address of an IP→Tel call (obtained from the Contact header in the INVITE message). **Note:** The source IP address can include the 'x' wildcard to represent single digits. For example: 10.8.8.xx represents all the addresses between 10.8.8.10 to 10.8.8.99. The '*' wildcard represents any number between 0 and 255, e.g., 10.8.8.* represents all addresses between 10.8.8.0 and 10.8.8.255.
Any SIP incoming call whose destination number matches the prefix defined in the 'Destination Phone Prefix' field *and* its source number matches the prefix defined in the adjacent 'Source Phone Prefix' field *and* its source IP address matches the address defined in the 'Source IP Address' field, is assigned to the trunk group entered in the field to the right of these fields. Note that IP to trunk group routing can be performed according to any combination of source / destination phone prefixes and source IP address, or using each independently. **Note:** For available notations that represent multiple numbers (used in the prefix columns), refer to Section 5.5.3.1 on page 67.	
Trunk Group ID	In each of the Trunk Group ID fields, enter the trunk group ID to which calls that match these prefixes are assigned.
Profile ID	Enter the number of the IP profile that is assigned to the routing rule.

5.5.5.3 Internal DNS Table

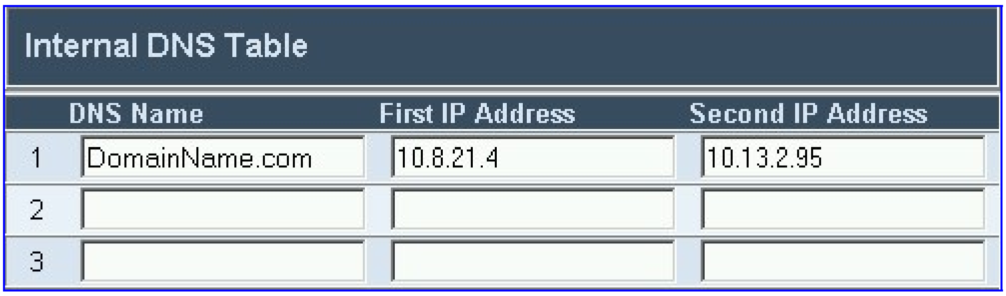
The internal DNS table, similar to a DNS resolution, translates hostnames into IP addresses. This table is used when hostname translation is required (e.g., 'Tel to IP Routing' table). Two different IP addresses can be assigned to the same hostname. If the hostname isn't found in this table, the gateway communicates with an external DNS server.

Assigning two IP addresses to hostname can be used for alternative routing (using the 'Tel to IP Routing' table).

➢ To configure the internal DNS table, take these 7 steps:

1. Open the 'Internal DNS Table' screen (**Protocol Management** menu > **Routing Tables** submenu > **Internal DNS Table** option); the 'Internal DNS Table' screen is displayed.

Figure 5-10: Internal DNS Table Screen



2. In the 'DNS Name' field, enter the hostname to be translated. You can enter a string up to 31 characters long.
3. In the 'First IP Address' field, enter the first IP address to which the hostname is translated.
4. In the 'Second IP Address' field, enter the second IP address to which the hostname is translated.
5. Repeat steps 2 to 4, for each Internal DNS Table entry.
6. Click the **Submit** button to save your changes.
7. To save the changes, refer to Section 5.9.2 on page 124.

5.5.5.4 Internal SRV Table

The Internal SRV table is used for resolving host names to DNS A-Records. Three different A-Records can be assigned to a hostname. Each A-Record contains the host name, priority, weight, and port.

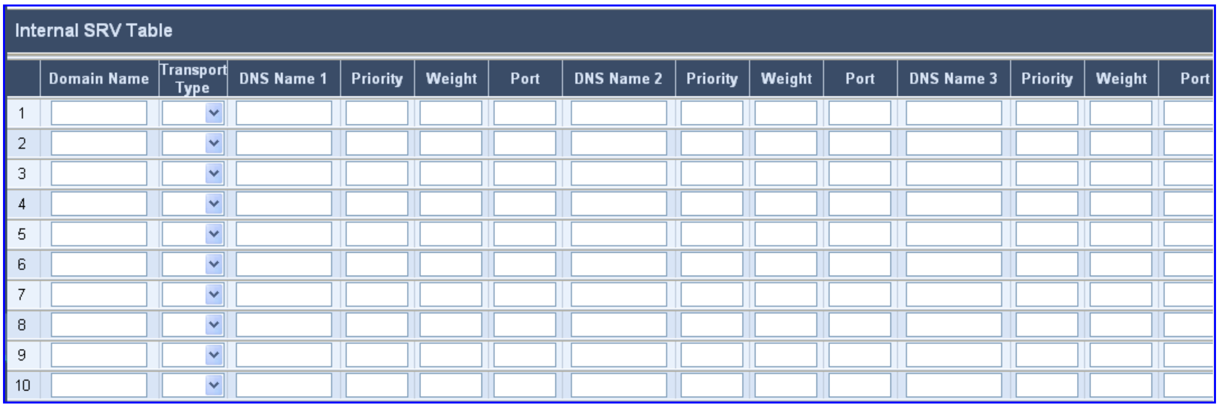
The *ini* file parameter SRV2IP is also used to configure the Internal SRV table and has the following format: <Internal Domain Name>, <Transport Type>, <DNS Name 1>, <Priority 1>, <Weight 1>, <Port 1>, <DNS Name 2>, <Priority 2>, <Weight 2>, <Port 2>, <DNS Name 3>, <Priority 3>, <Weight 3>, <Port 3>. This parameter can appear up to 10 times.

Note: If the Internal SRV table is configured, the gateway first tries to resolve a domain name using this table. If the domain name isn't found, the gateway performs an SRV resolution using an external DNS server.

➢ To configure the Internal SRV table, take these 9 steps:

1. Open the 'Internal SRV Table' screen (**Protocol Management** menu > **Routing Tables** submenu > **Internal SRV Table** option); the 'Internal DNS Table' screen is displayed.

Figure 5-11: Internal SRV Table Screen



2. In the 'Domain Name' field, enter the hostname to be translated. You can enter a string up to 31 characters long.
3. From the 'Transport Type' drop-down list, select a transport type.
4. In the 'DNS Name 1' field, enter the first DNS A-Record to which the hostname is translated.
5. In the 'Priority', 'Weight' and 'Port' fields, enter the relevant values
6. Repeat steps 4 to 5, for the second and third DNS names, if required.
7. Repeat steps 2 to 6, for each Internal SRV Table entry.
8. Click the **Submit** button to save your changes.
9. To save the changes so they are available after a hardware reset or power fail, refer to Section 5.9.2 on page 124.

5.5.5.5 Reasons for Alternative Routing

The Reasons for Alternative Routing screen includes two tables (Tel→IP and IP→Tel). Each table enables you to define up to 4 different release reasons. If a call is released as a result of one of these reasons, the gateway tries to find an alternative route to that call. The release reason for IP→Tel calls is provided in Q.931 notation. The release reason for Tel→IP calls is provided in SIP 4xx, 5xx and 6xx response codes. For Tel→IP calls an alternative IP address, for IP→Tel calls an alternative trunk group.

Refer to 'Tel to IP Routing Table' on page 70 for information on defining an alternative IP address. Refer to the 'IP to Trunk Group Routing Table' on page 72 for information on defining an alternative trunk group.

You can use this table for example:

For Tel→IP calls, when there is no response to an INVITE message (after INVITE retransmissions), and the gateway then issues an internal 408 'No Response' implicit release reason.

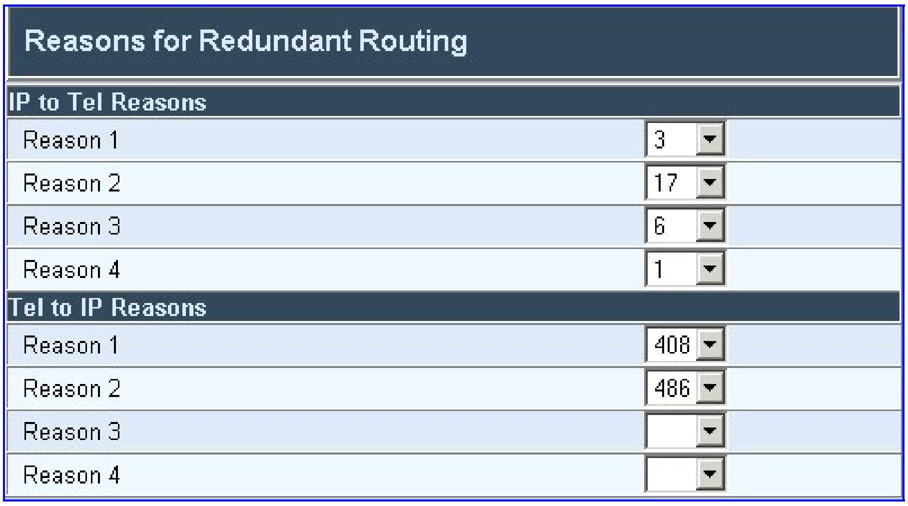
For IP→Tel calls, when the destination is busy, and release reason #17 is issued or for other call releases that issue the default release reason (#3). Refer to 'DefaultReleaseCause' in Table 6-7.

Note: The reasons for alternative routing option for Tel→IP calls only apply when Proxy isn't used.

➢ **To configure the reasons for alternative routing, take these 5 steps:**

1. Open the 'Reasons for Alternative Routing' screen (**Protocol Management** menu > **Routing Tables** submenu > **Reasons for Alternative Routing** option); the 'Reasons for Alternative Routing' screen is displayed.

Figure 5-12: Reasons for Alternative Routing Screen



2. In the 'IP to Tel Reasons' table, from the drop-down list select up to 4 different call failure reasons that invoke an alternative IP to Tel routing.
3. In the 'Tel to IP Reasons' table, from the drop-down list select up to 4 different call failure reasons that invoke an alternative Tel to IP routing.
4. Click the **Submit** button to save your changes.
5. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

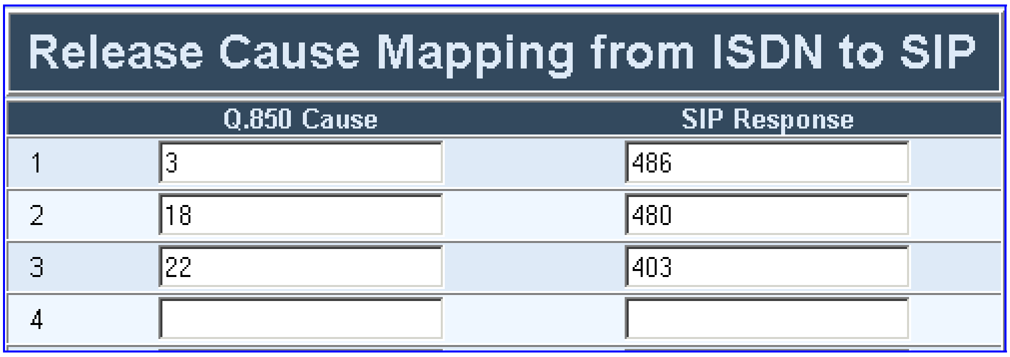
5.5.5.6 Release Cause Mapping

The Release Cause Mapping screen enables the gateway to map (up to 12) different SIP Responses to Q.850 Release Causes and vice versa, thereby overriding the hard-coded mapping mechanism (described in Appendix H on page 379).

➢ **To configure the release cause mapping, take these 5 steps:**

1. Open the ‘Release Cause Mapping’ screen (**Protocol Management** menu > **Routing Tables** submenu > **Release Cause Mapping** option); the ‘Release Cause Mapping’ screen is displayed.

Figure 5-13: Release Cause Mapping from ISDN to SIP



2. In the ‘Release Cause Mapping from ISDN to SIP’ table, define (up to 12) different Q.850 Release Causes and the SIP Responses they are mapped to.
3. In the ‘Release Cause Mapping from SIP to ISDN’ table, define (up to 12) different SIP Responses and the Q.850 Release Causes they are mapped to.
4. Click the **Submit** button to save your changes.
5. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

5.5.6 Configuring the Profile Definitions

Utilizing the Profiles feature, the gateway provides high-level adaptation when connected to a variety of equipment (from both Tel and IP sides) and protocols, each of which requires a different system behavior. Using Profiles, users can assign different Profiles (behavior) on a per-call basis, using the Tel to IP and IP to Trunk Group Routing tables, or associate different Profiles to the gateway’s B-channels(s). The Profiles contain parameters such as Coders, T.38 Relay, Voice and DTMF Gains, Silence Suppression, Echo Canceler, RTP DiffServ and more. The Profiles feature allows users to tune these parameters or turn them on or off, per source or destination routing and/or the specific gateway or its ports. For example, specific E1/T spans can be designated for to have a profile which always uses G.711.

Each call can be associated with one or two Profiles, Tel Profile and (or) IP Profile. If both IP and Tel profiles apply to the same call, the coders and other common parameters of the preferred Profile (determined by the Preference option) are applied to that call. If the Preference of the Tel and IP Profiles is identical, the Tel Profile parameters are applied.

Note: The default values of the parameters in the Tel and IP Profiles are identical to the Web/*ini* file parameter values. If a value of a parameter is changed in the Web/*ini* file, it is automatically updated in the Profiles correspondingly. After any parameter in the Profile is modified by the user, modifications to parameters in the Web/*ini* file no longer impact that Profile.

5.5.6.1 Coder Group Settings

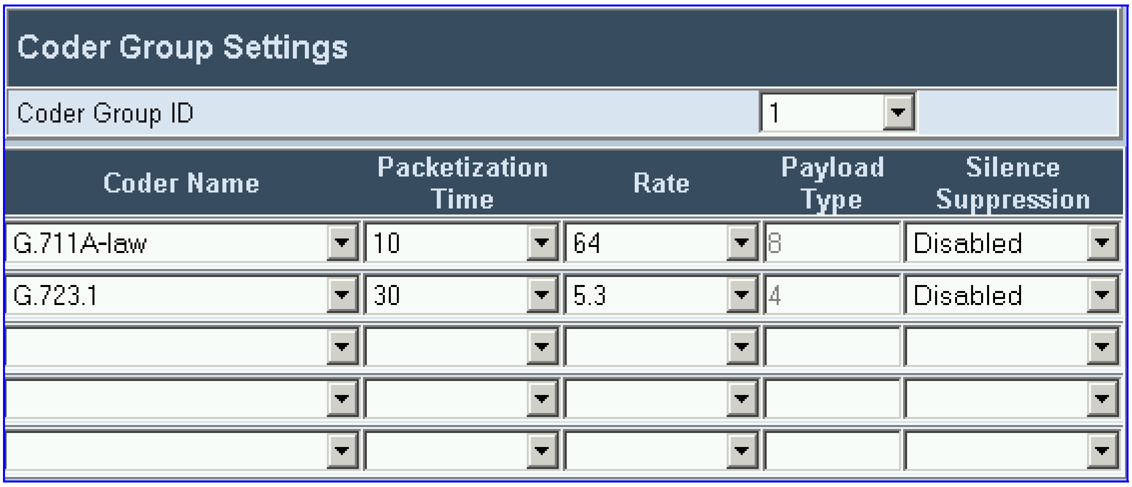
Use the Coder Group Settings screen to define up to four different coder groups. These coder groups are used in the Tel and IP Profile Settings screens to assign different coders to Profiles.

For each group you can define the first to fifth preferred coders (and their attributes) for the gateway. The first coder is the highest priority coder and is used by the gateway whenever possible. If the far end gateway cannot use the coder assigned as the first coder, the gateway attempts to use the next coder and so forth.

➢ To configure the coder group settings, take these 11 steps:

1. Open the 'Coder Group Settings' screen (**Protocol Management** menu > **Profile Definitions** submenu > **Coder Group Settings** option); the 'Coder Group Settings' screen is displayed.

Figure 5-14: Coder Group Settings Screen



2. From the Coder Group ID drop-down list, select the coder group you want to edit (up to four coder groups can be configured).
3. From the Coder Name drop-down list, select the coder you want to use. For the full list of available coders and their corresponding attributes, refer to the *ini* file parameter 'CoderName_ID' (described in Table 6-7).

   Note: Each coder can appear only once.
4. From the Packetization Time drop-down list, select the packetization time (in msec) for the coder you selected. The packetization time determines how many coder payloads are combined into a single RTP packet.

   Note 1: If not specified, the ptime gets a default value.
   Note 2: The ptime specifies the packetization time the gateway expects to receive. The gateway always uses the ptime requested by the remote side for sending RTP packets.
5. From the Rate drop-down list, select the bit rate (in kbps) for the coder you selected.
6. In the Payload Type field, if the payload type for the coder you selected is dynamic, enter a value from 0 to 120 (payload types of 'well-known' coders cannot be modified). The payload type identifies the format of the RTP payload.
   Note: If not specified, a default is used.
7. From the Silence Suppression drop-down list, enable or disable the silence suppression option for the coder you selected.
   Note: For G.729 it is also possible to select silence suppression without adaptations.
8. Repeat steps 3 to 7 for the second to fifth coders (optional).
9. Repeat steps 2 to 8 for the second to fourth coder groups (optional).

10. Click the **Submit** button to save your changes.
11. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

Notes:

- Only the ptime of the first coder in the defined coder list is declared in INVITE / 200 OK SDP, even if multiple coders are defined.
- If the coder G.729 is selected and silence suppression is enabled (for this coder), the gateway includes the string ‘annexb=no’ in the SDP of the relevant SIP messages. If silence suppression is set to ‘Enable w/o Adaptations’, ‘annexb=yes’ is included. An exception to this logic is when the remote gateway is a Cisco device (IsCiscoSCEMode).

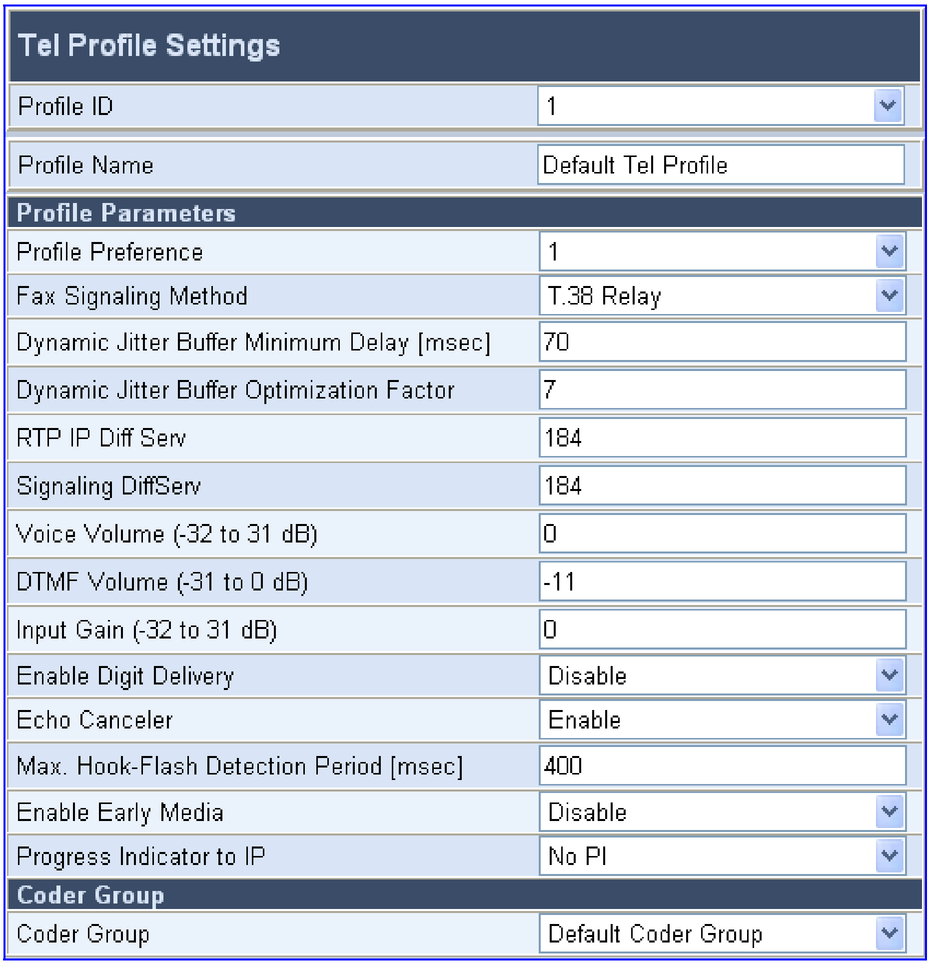
5.5.6.2 Tel Profile Settings

Use the Tel Profile Settings screen to define up to four different Tel Profiles. These Profiles are used in the ‘Trunk Group’ table to associate different Profiles to gateway’s B-channels, thereby applying different behavior to different gateway's B-channels.

➢ **To configure the Tel Profile settings, take these 8 steps:**

1. Open the ‘Tel Profile Settings’ screen (**Protocol Management** menu > **Profile Definitions** submenu > **Tel Profile Settings** option); the ‘Tel Profile Settings’ screen is displayed.

Figure 5-15: Tel Profile Settings Screen



2. In the ‘Profile ID’ drop-down list, select the Tel Profile you want to edit (up to four Tel Profiles can be configured).
3. In the ‘Profile Preference’ drop-down list, select the preference (1-10) of the current Profile. The preference option is used to determine the priority of the Profile. Where ‘20’ is the highest preference value. If both IP and Tel profiles apply to the same call, the coders and other common parameters (noted by an asterisk in the description of the parameter TelProfile_ID) of the preferred Profile are applied to that call. If the Preference of the Tel and IP Profiles is identical, the Tel Profile parameters are applied.
   Note: If the coder lists of both IP and Tel Profiles apply to the same call, an intersection of the coders is performed (i.e., only common coders remain). The order of the coders is determined by the preference.
4. Configure the Profile’s parameters according to your requirements. For detailed information on each parameter refer to the description of the screen in which it is configured as an individual parameter.
5. In the ‘Coder Group’ drop-down list, select the coder group you want to assign to that Profile. You can select the gateway’s default coders (refer to Section 5.5.1.1 on page 63) or one of the coder groups you defined in the Coder Group Settings screen (refer to Section 5.5.6.1 on page 78).
6. Repeat steps 2 to 6 for the second to fifth Tel Profiles (optional).
7. Click the **Submit** button to save your changes.
8. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

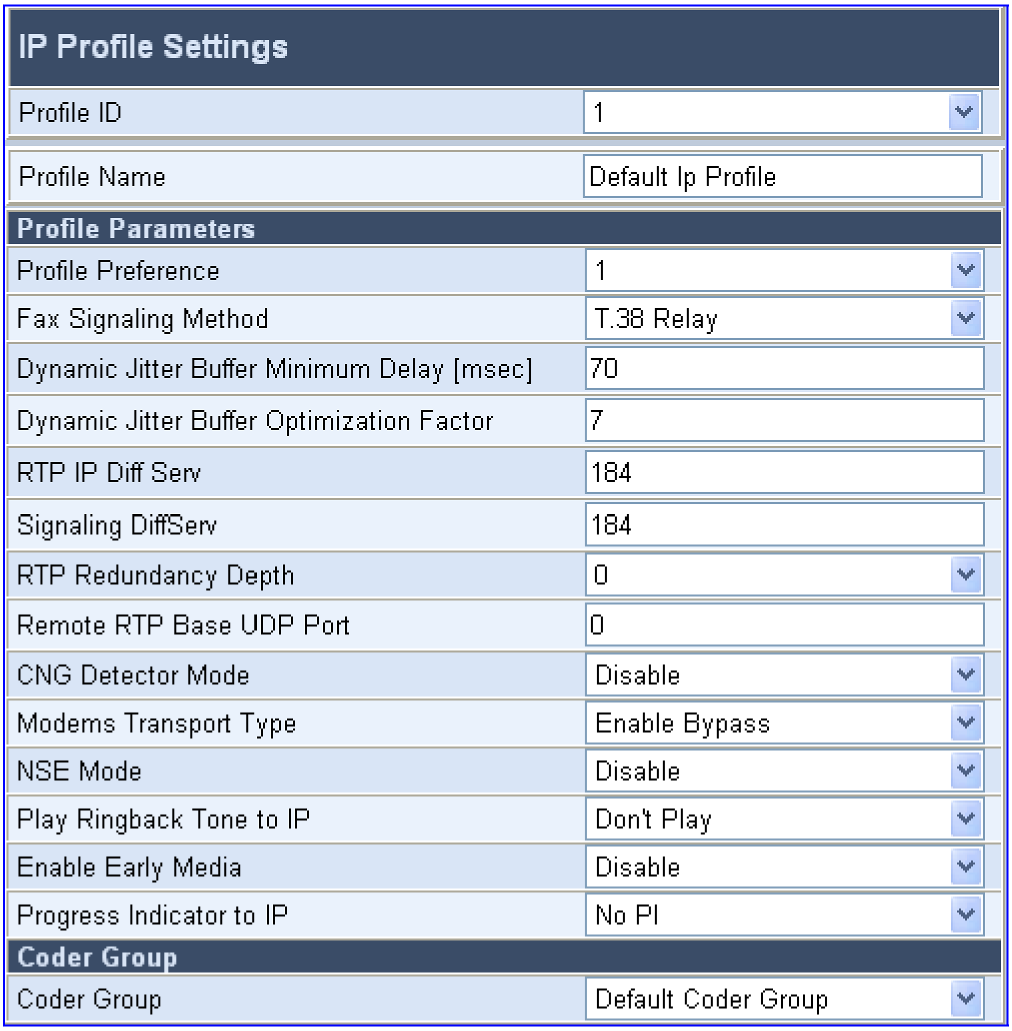
5.5.6.3 IP Profile Settings

Use the IP Profile Settings screen to define up to four different IP Profiles. These Profiles are used in the Tel to IP and IP to Trunk Group Routing tables to associate different Profiles to routing rules. IP Profiles can also be used when working with Proxy server (set ‘AlwaysUseRouteTable’ to 1).

➢ To configure the IP Profile settings, take these 8 steps:

1. Open the ‘IP Profile Settings’ screen (**Protocol Management** menu > **Profile Definitions** submenu > **IP Profile Settings** option); the ‘IP Profile Settings’ screen is displayed.

Figure 5-16: IP Profile Settings Screen



2. In the 'Profile ID' drop-down list, select the IP Profile you want to edit (up to four IP Profiles can be configured).
3. In the 'Profile Preference' drop-down list, select the preference (1-10) of the current Profile. The preference option is used to determine the priority of the Profile. Where '20' is the highest preference value. If both IP and Tel profiles apply to the same call, the coders and other common parameters (noted by an asterisk in the description of the parameter IPProfile_ID) of the preferred Profile are applied to that call. If the Preference of the Tel and IP Profiles is identical, the Tel Profile parameters are applied.
   Note: If the coder lists of both IP and Tel Profiles apply to the same call, an intersection of the coders is performed (i.e., only common coders remain). The order of the coders is determined by the preference.
4. Configure the Profile's parameters according to your requirements. For detailed information on each parameter refer to the description of the screen in which it is configured as an individual parameter.
5. In the 'Coder Group' drop-down list, select the coder group you want to assign to that Profile. You can select the gateway's default coders (refer to Section 5.5.1.1 on page 63) or one of the coder groups you defined in the Coder Group Settings screen (refer to Section 5.5.6.1 on page 78).
6. Repeat steps 2 to 6 for the second to fifth IP Profiles (optional).
7. Click the **Submit** button to save your changes.
8. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

5.5.7 Configuring the Trunk Group Table

Use the Trunk Group table to assign trunk groups, profiles and logical telephone numbers to the gateway's E1/T1 B-channels. Trunk Groups are used for routing IP→Tel calls with common rules. Channels that are not defined are disabled.

➢ To configure the Trunk Group table, take these 4 steps:

1. Open the ‘Trunk Group Table’ screen (**Protocol Management** menu > **Trunk Group**); the ‘Trunk Group Table’ screen is displayed.

Figure 5-17: Trunk Group Table Screen

	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	Profile ID
1	1	2	*	6000	1	2
2	3	3	1-25	7000	2	0
3	3	3	26-30	8000	3	1
4						
5						

2. Configure the Trunk Group according to Table 5-8.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

Table 5-8: Trunk Group Table

Parameter	Description
From Trunk	Starting physical trunk number (0 to 7).
To Trunk	Ending physical trunk number (0 to 7).
Channels	To enable the trunk’s B-channels, you **must** enter their number in this field. [n-m] represents a range of channels. For example, enter [1-24] to specify the channels from 1 to 24. **Note 1:** The number of defined channels must not exceed the number of the trunk’s B-channels (1-24 for T1 spans and 1-31 for E1 spans). **Note 2:** To represent all B-channels use a single asterisk instead.
Phone Number	In each of the Phone Number fields, enter the first number in an ordered sequence that is assigned to the range of channels defined in the adjacent ‘Channels’ field. **Note:** This field is optional. The logical numbers defined in this field are used when an incoming PSTN / PBX call doesn’t contain the calling number or called number (the latter being determined by the parameter ‘ReplaceEmptyDstWithPortNumber’); these numbers are used to replace them. These logical numbers are also used for B-channel allocation for IP to Tel calls, if the trunk group’s ‘Channel Select Mode’ is set to ‘By Phone Number’.
Trunk Group ID	In each of the Trunk Group ID fields, enter the trunk group ID (1-99) assigned to the channels. The same trunk group ID can be used for more than one group of channels. Trunk group ID is used to define a group of common behavior channels that are used for routing IP to Tel calls. If an IP to Tel call is assigned to a trunk group, the call is routed to the channel or channels that correspond to the trunk group ID. You can configure the Trunk Group Settings table to determine the method in which new calls are assigned to channels within the trunk groups (refer to Section 5.5.8 on page 83). **Note:** You must configure the IP to Trunk Group Routing Table (assigns incoming IP calls to the appropriate trunk group). If you do not configure the IP to Trunk Group Routing Table, calls do not complete. For information on how to configure this table, refer to Section 5.5.5.2 on page 72.
Profile ID	Enter the number of the Tel profile that is assigned to the B-channels defined in the ‘Channels’ field.

5.5.8 Configuring the Trunk Group Settings

The Trunk Group Settings Table is used to determine the method in which new calls are assigned to B-channels within each trunk group. If such a rule doesn't exist (for a specific Trunk group), the global rule, defined by the Channel Select Mode parameter (Protocol Definition > General Parameters), applies.

➢ To configure the Trunk Group Settings table, take these 7 steps:

1. Open the 'Trunk Group Settings' screen (**Protocol Management** menu > **Trunk Group Settings**); the 'Trunk Group Settings' screen is displayed.

Figure 5-18: Trunk Group Settings Screen

	Trunk Group ID	Channel Select Mode
1	1	Cyclic Ascending
2	2	Ascending
3	3	Descending
4		
5		
6		
7		
8		
9		
10		
11		
12		

2. In the **Routing** Index drop-down list, select the range of entries that you want to edit (up to 24 entries can be configured).
3. In the **Trunk Group ID** field, enter the Trunk Group ID number.
4. In the **Channel Select Mode** drop-down list, select the Channel Select Mode that determines the method in which new calls are assigned to B-channels within the Trunk groups entered in the field to the right of this field. For information on available Channel Select Modes, refer to Table 5-9.
5. Repeat steps 4 and 5, for each defined Trunk group.
6. Click the **Submit** button to save your changes.
7. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

Table 5-9: Channel Select Modes

Mode	Description
By phone number	Select the gateway port according to the called number (refer to the note below).
Cyclic Ascending	Select the next available channel in an ascending cycle order. Always select the next higher channel number in the Trunk Group. When the gateway reaches the highest channel number in the Trunk Group, it selects the lowest channel number in the Trunk Group and then starts ascending again (default).
Ascending	Select the lowest available channel. Always start at the lowest channel number in the Trunk Group and if that channel is not available, select the next higher channel.
Cyclic Descending	Select the next available channel in descending cycle order. Always select the next lower channel number in the Trunk Group. When the gateway reaches the lowest channel number in the Trunk Group, it selects the highest channel number in the Trunk Group and then start descending again.
Descending	Select the highest available channel. Always start at the highest channel number in the Trunk Group and if that channel is not available, select the next lower channel.
Number + Cyclic Ascending	First select the gateway port according to the called number (refer to the note below). If the called number isn't found, then select the next available channel in ascending cyclic order. Note that if the called number is found, but the port associated with this number is busy, the call is released.

Note: The internal numbers of the gateway's B-channels are defined in the 'Trunk Group Table' under the 'Phone Number' column. For detailed information on the 'Trunk Group Table', refer to Section 5.5.7 on page 82).

5.6 Advanced Configuration

Use this menu to set the gateway's advanced configuration parameters (for advanced users only).

5.6.1 Configuring the Network Settings

From the Network Settings you can:

- Configure the IP Settings.
- Configure the Application Settings.
- Configure the NFS Settings (refer to Section 5.6.1.1 below).
- Configure the IP Routing Table (refer to Section 5.6.1.2 on page 86).
- Configure the VLAN Settings.

5.6.1.1 Configuring the NFS Settings

Network File System (NFS) enables the gateway to access a remote server's shared files and directories and to handle them as if they're located locally. A file system, the NFS is independent of machine types, OSs, and network architectures. Up to five different NFS file systems can be configured.

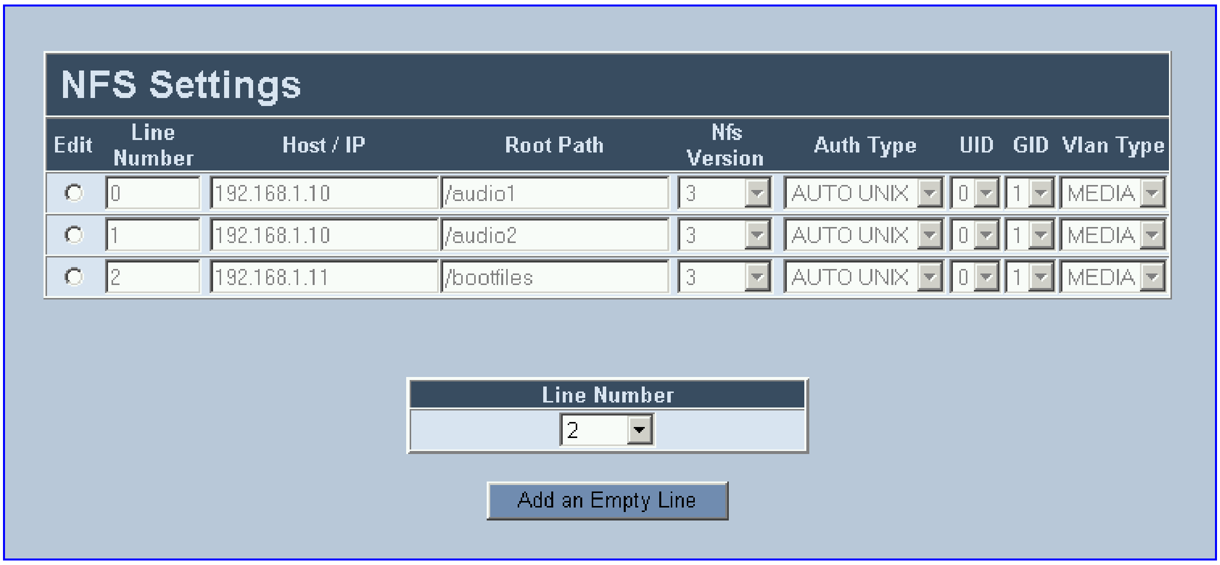
NFS is utilized by the gateway to load the *cmp*, *ini* and configuration files via the Automatic Update mechanism (refer to Section 11.3 on page 249).

Note that an NFS file server can share multiple file systems. There must be a separate row for each remote file system shared by the NFS file server that needs to be accessed by the gateway.

➢ To configure the NFS Settings parameters, take these 7 steps:

1. Open the 'Application Settings' screen (**Advanced Configuration** menu > **Network Settings** > **Application Settings** option); the 'Application Settings' screen is displayed.
2. Open the NFS Table screen by clicking the arrow sign (-->) to the right of the NFS label; the NFS Table screen is displayed (Figure 5-19).

Figure 5-19: NFS Settings Table Screen



3. To add a remote NFS file system, select an available line number from the **Line Number** drop-down list.
4. Click the **Add an Empty Line** button; an empty line appears.
5. Configure the NFS Settings. The NFS parameters are described in Table 6-1 on page 130.
6. Click the **Apply New Settings** button; the remote NFS file system is mounted immediately. Check the Syslog server for the 'NFS mount was successful' message.
7. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

Note: To avoid terminating calls in progress, a row must not be deleted or modified while the board is currently accessing files on that remote NFS file system.

➢ To delete a remote NFS file system, take these 3 steps:

1. Click the **Edit** radio button for the row to be deleted.
2. Click the **Delete Line** button; the row is deleted.
3. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

➢ To modify an existing remote NFS file system, take these 4 steps:

1. Click the **Edit** radio button for the row to be modified.
2. Change the values on the selected row according to your requirements.
3. Click the **Apply New Settings** button; the remote NFS file system is mounted using the new settings. Check the Syslog server for the 'NFS mount was successful' message.
4. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

Figure 5-20 below shows an example of an NFS table definition via *ini* file using parameter tables (for information on parameter tables, refer to Section 11.5 on page 253).

Figure 5-20: NFS *ini* File Example

```
[ NFSServers ]
FORMAT NFSServers_Index = NFSServers_HostOrIP, NFSServers_RootPath,
NFSServers_NfsVersion, NFSServers_AuthType, NFSServers_UID, NFSServers_GID,
NFSServers_VlanType;
NFSServers 1 = 101.1.13, /audio1, 3, 1, 0, 1, 1;
[ \NFSServers ]
```

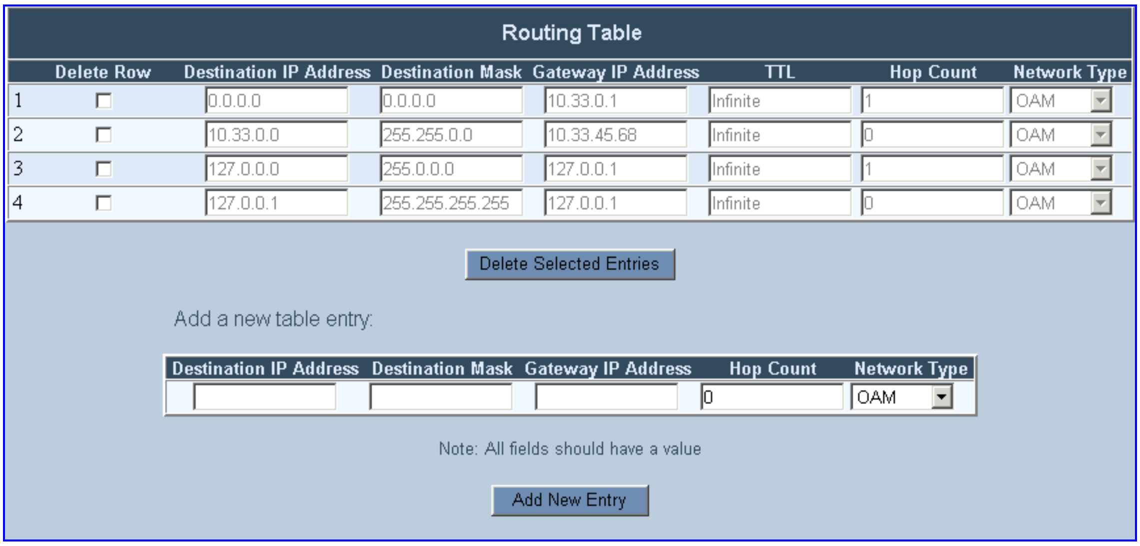
5.6.1.2 Configuring the IP Routing Table

The IP routing table is used by the gateway to determine IP routing rules. It can be used, for example, to define static routing rules for the OAM and Control networks since a default gateway isn't supported for these networks (refer to Section 9.10.1 on page 237). Before sending an IP packet, the gateway searches this table for an entry that matches the requested destination host / network. If such entry is found, the gateway sends the packet to the indicated router. If no explicit entry is found, the packet is sent to the default gateway (configured in Network Settings>IP Settings screen). Up to 50 routing entries are available.

➢ To configure the IP Routing table, take these 3 steps:

1. Open the 'IP Routing Table' screen (**Advanced Configuration** menu > **Network Settings** > **IP Routing Table** option); the 'IP Routing Table' screen is displayed.

Figure 1-3: IP Routing Table Screen



Routing Table

	Delete Row	Destination IP Address	Destination Mask	Gateway IP Address	TTL	Hop Count	Network Type
1		0.0.0.0	0.0.0.0	10.33.0.1	Infinite	1	OAM
2		10.33.0.0	255.255.0.0	10.33.45.68	Infinite	0	OAM
3		127.0.0.0	255.0.0.0	127.0.0.1	Infinite	1	OAM
4		127.0.0.1	255.255.255.255	127.0.0.1	Infinite	0	OAM

Delete Selected Entries

Add a new table entry:

Destination IP Address	Destination Mask	Gateway IP Address	Hop Count	Network Type
			0	OAM

Note: All fields should have a value

Add New Entry

2. Use the 'Add a new table entry' pane to add a new routing rule. Each field in the IP routing table is described in Table 5-10.
3. Click the button **Add New Entry**; the new routing rule is added to the IP routing table.

Table 5-10: IP Routing Table Column Description

<table>
<tr><th>Column Name
[ini File Parameter Name]</th><th>Description</th></tr>
<tr><td>Delete Row</td><td>To delete IP routing rules from the IP Routing Table, check the Delete Row checkbox in the rows of the routing rules you want to delete and click the button Delete Selected Entries; the routing rules are removed from the table.</td></tr>
<tr><td>Destination IP Address</td><td>Specifies the IP address of the destination host / network.</td></tr>
<tr><td>Destination Mask</td><td>Specifies the subnet mask of the destination host / network.</td></tr>
<tr><td colspan="2">The address of the host / network you want to reach is determined by an AND operation that is applied on the fields ‘Destination IP Address’ and ‘Destination Mask’.
For example:
To reach the network 10.8.x.x, enter 10.8.0.0 in the field ‘Destination IP Address’ and 255.255.0.0 in the field ‘Destination Mask’. As a result of the AND operation, the value of the last two octets in the field ‘Destination IP Address’ is ignored.
To reach a specific host, enter its IP address in the field ‘Destination IP Address’ and 255.255.255.255 in the field ‘Destination Mask’.</td></tr>
<tr><td>Gateway IP Address</td><td>Specifies the IP address of the router to which the packets are sent if their destination matches the rules in the adjacent columns.</td></tr>
<tr><td>TTL</td><td>A read-only field that indicates the time period for which the specific routing rule is valid. The lifetime of a static route is infinite.</td></tr>
<tr><td>Hop Count</td><td>The maximum number of allowed routers between the gateway and destination.</td></tr>
<tr><td>Network Type</td><td>Specifies the network type the routing rule is applied to.
OAM (default).
Control.
Media.
For detailed information on the network types, refer to Section 9.10.1 on page 237.</td></tr>
</table>

5.6.2 Configuring the Media Settings

Use these menus to set the gateway’s channel parameters. These parameters are applied to all the gateway's channels. Several Channels Settings parameters can be configured per call using profiles (refer to Section 5.5.5.6 on page 77). Note that channel parameters are changeable on-the-fly. Changes take effect from the next call.

From the Media Settings you can:

- Define the Voice Settings.
- Define the Fax / Modem and CID Settings.
- Define the RTP / RTCP Settings.
- Define the IPmedia Settings.
- Define General Media Settings.

Notes:

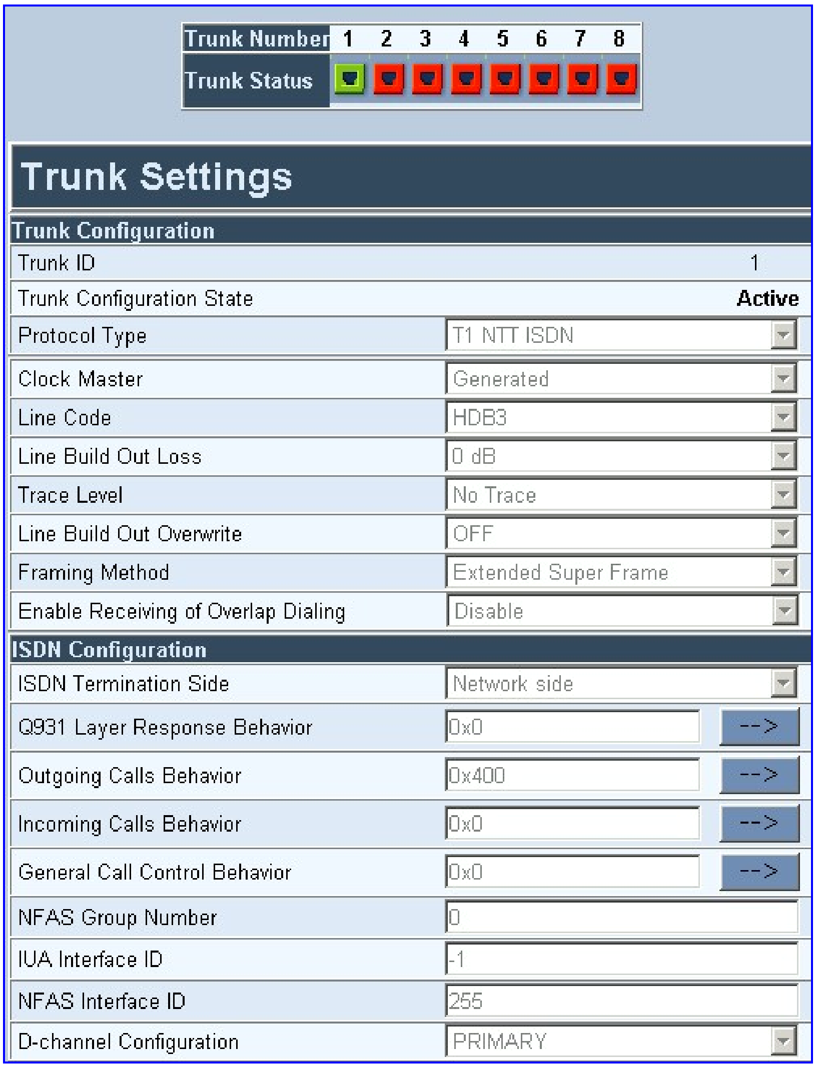
- The parameters ‘MF Transport Type’ (Voice Settings screen) and the five Answer Detector (IPmedia Settings screen) parameters are not applicable to the the gateway.
- The parameters ‘Fax Transport Mode’ (Fax / Modem / CID Settings screen) is overridden by the parameter ‘IsFaxUsed’.

5.6.3 Configuring the Trunk Settings

➢ **To configure the Trunk Settings, take these 10 steps:**

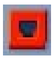
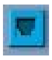
1. Open the 'Trunk Settings' screen (**Advanced Configuration** menu > **Trunk Settings**); the 'Trunk Settings' screen is displayed. Initially, the screen appears with the parameters fields grayed (indicating read-only). The **Stop Trunk** button appears at the bottom of the screen. The Trunk Status indicators appear colored. Table 5-11 shows the possible indicators and their descriptions.

Figure 5-21: E1/T1 Trunk Settings Screen



2. To configure the parameters of a specific trunk, from the trunks displayed on the top, select the trunk you want to configure by clicking the Trunk's Status indicator. The first parameter named 'Trunk ID' changes according to the trunk you click. The parameters displayed are for the selected trunk only.

Table 5-11: Trunks Status Color Indicator Keys

Indicator	Color	Description
	Gray	Disabled
	Green	Active-OK
	Yellow	RAI Alarm
	Red	LOS/LOF Alarm
	Blue	AIS Alarm
	Orange	D-channel Alarm (ISDN only)

3. To modify the selected trunk's parameters, click the **Stop Trunk** button; the trunk is stopped, the status of the parameter 'Trunk Configuration State' changes to 'Non Active', the parameters are no longer grayed and can be modified and the **Apply Trunk Settings** button appears at the bottom of the screen. When all trunks are stopped, the **Apply to all Trunks** button also appears at the bottom of the screen.

> **Note:** If the trunk can't be stopped because it provides the gateway's clock (assuming the the gateway is synchronized with the E1/T1 clock), assign a different E1/T1 trunk to provide the gateway's clock or enable 'TDM Bus PSTN Auto Clock' on the TDM Bus Settings screen.
> To assign a different E1/T1 trunk that provides the gateway's clock, access the 'TDM Bus Setting' screen and change the 'TDM Bus Local Reference' number to any other trunk number (this operation can be performed on-the-fly).

4. Select the 'Protocol Type' you use. Note that different trunks can be defined with different protocols (CAS or ISDN variants) on the same gateway (subject to the constraints in the Mediant & TP Series SIP Digital Gateways Release Notes).

> **Note:** When modifying the 'Protocol Type' field, the menu is automatically updated according to the selected protocol (ISDN, CAS or Transparent). Additional parameters are appropriate to the selected protocol type.

5. Modify the relevant trunk configuration parameters according to your requirements.
6. To configure the different behavior bits: either enter the exact hexadecimal value of the bits in the field to the right of the relevant behavior parameter, or directly configure each bit field by completing the following steps:
   - Click the arrow button (-->) to the right of the relevant behavior parameter; a new window appears.
   - Modify each bit field according to your requirements.
   - Click the **Submit** button to save your changes.

7. After modifying the parameters:
   - To apply the changes to the selected trunk only, click the **Apply Trunk Settings** button.
   - To apply the changes to all the trunks, click the **Apply to all Trunks** button.
8. The screen is refreshed; parameters become read-only (indicated by being grayed). The **Stop Trunk** button appears at the bottom of the screen.
9. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

> **Note:** Some parameter configuration options require a device reset; when this is the case, the Web Interface prompts the user.

10. To reset the the gateway, refer to Section 5.9.3 on page 125.

5.6.4 Configuring SS7 Tunneling

For a detailed description of the SS7 Tunneling parameters, refer to Section 12.2.4 on page 272.

From the SS7 Configuration menu you can:

- Configure M2P2 Attributes (refer to Section 5.6.4.1)
- Configure Links (refer to Section 5.6.4.2 on page 91)
- Configure Sigtran Group IDs (refer to Section 5.6.4.3 on page 93)
- Configure Sigtran Interface IDs (refer to Section 5.6.4.4 on page 94)

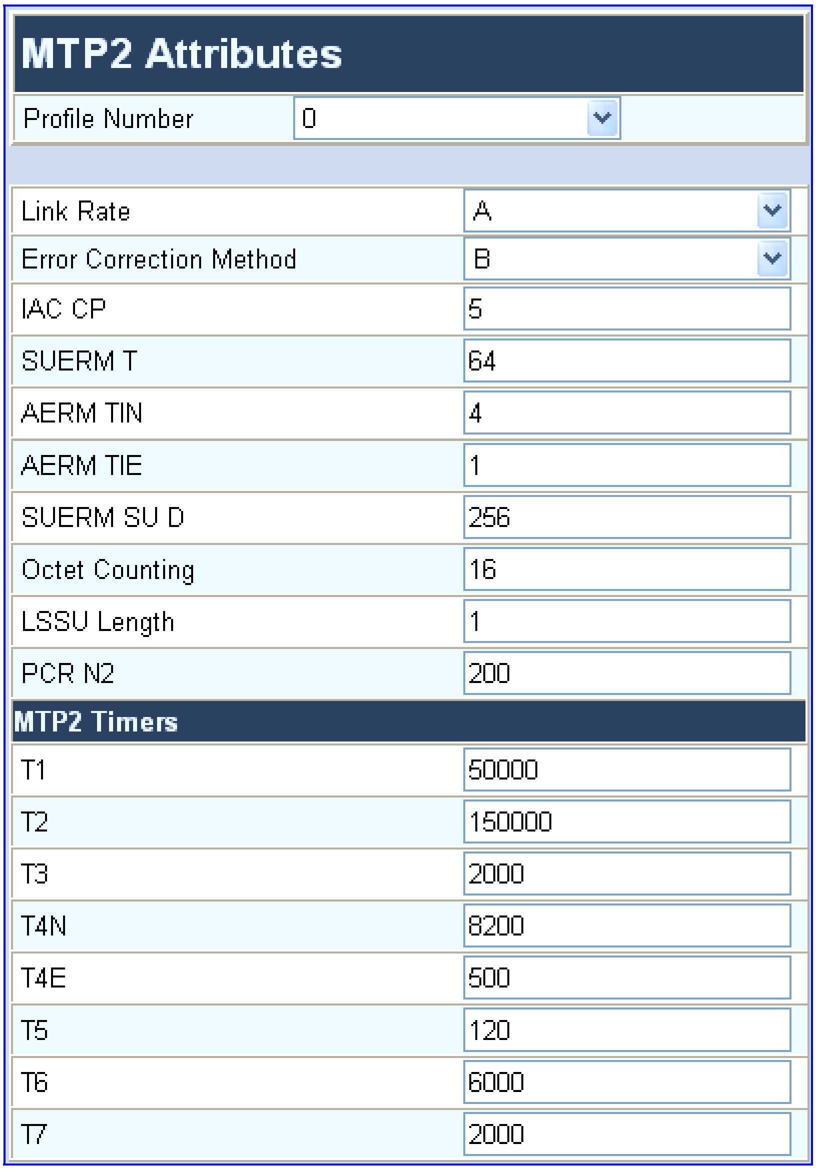
5.6.4.1 Configuring M2P2 Attributes

For a detailed description of M2P2 attributes, refer to Section 12.2.4 on page 272.

➢ **To configure the M2P2 Attributes parameters, take these 4 steps:**

1. Open the ‘M2P2 Attributes‘ screen (**Advanced Configuration** menu > **SS7 Configuration** > **M2P2 Attributes** option); the ‘M2P2 Attributes’ screen is displayed.

Figure 5-22: M2P2 Attributes Screen



2. **Configure** or modify the parameters as desired (for a description of the parameters, refer to Section 12.2.4 on page 272).
3. Click **Apply**.
4. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

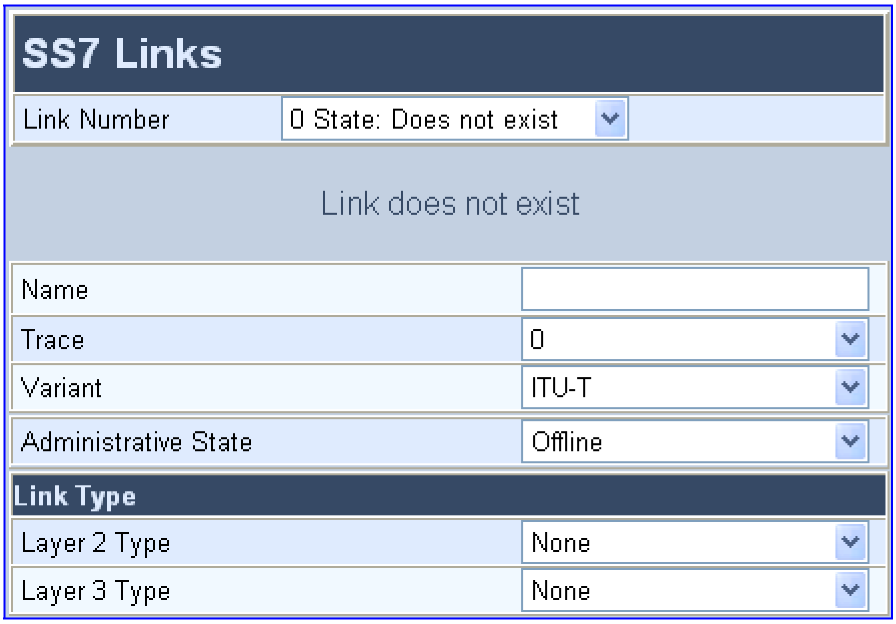
5.6.4.2 Configuring Links

For a detailed description of Links, refer to Section 12.2.4 on page 272.

➢ To configure the Links parameters, take these 4 steps:

1. Open the 'Links' screen (**Advanced Configuration** menu > **SS7 Configuration** > **Links** option); the 'Links' screen is displayed.

Figure 5-23: Links Screen



2. Configure or modify the parameters as desired (for a description of the parameters, refer to Section 12.2.4 on page 272).
3. Click **Create**.
4. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

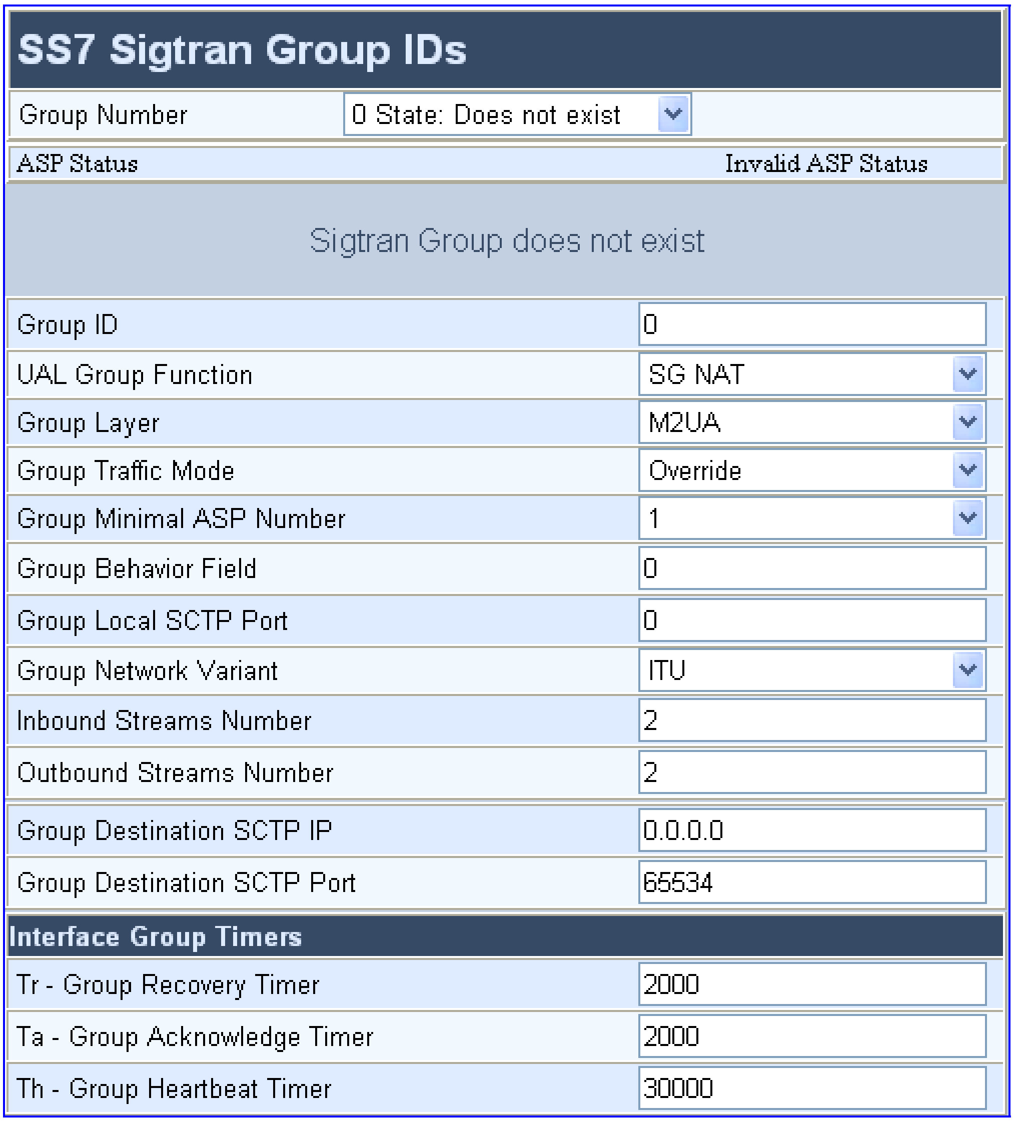
5.6.4.3 Configuring Sigtran Group IDs

For a detailed description of Sigtran Group IDs, refer to Section 12.2.4 on page 272.

➢ **To configure the Sigtran Group IDs parameters, take these 4 steps:**

1. Open the 'Sigtran Group IDs' screen (**Advanced Configuration** menu > **SS7 Configuration** > Sigtran Group IDs option); the 'Sigtran Group IDs' screen is displayed.

Figure 5-24: Sigtran Group IDs Screen



2. **Configure** or modify the parameters as desired (for a description of the parameters, refer to Section 12.2.4 on page 272).
3. Click **Create**.
4. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

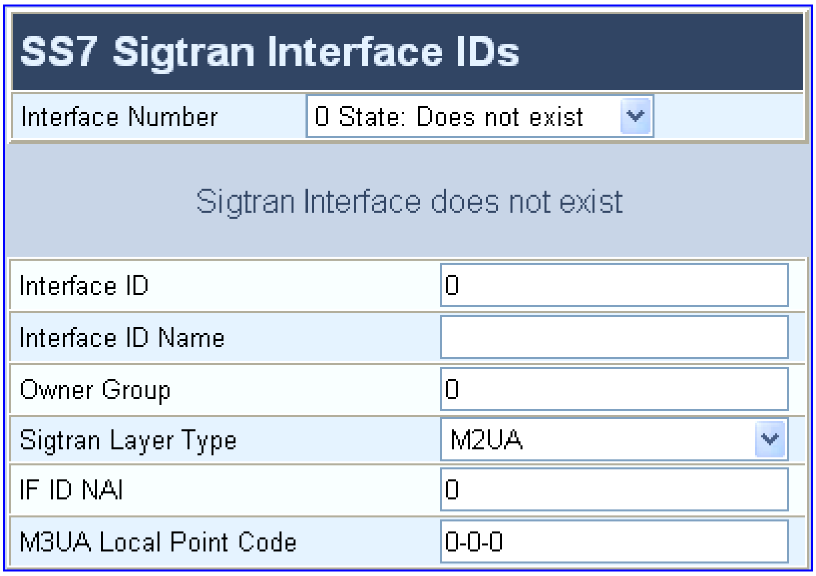
5.6.4.4 Configuring Sigtran Interface IDs

For a detailed description of Sigtran Interface IDs, refer to Section 12.2.4 on page 272.

➢ **To configure the Sigtran Interface IDs parameters, take these 4 steps:**

1. Open the 'Sigtran Interface IDs' screen (**Advanced Configuration** menu > **SS7 Configuration** > Sigtran Interface IDs option); the 'Sigtran Interface IDs' screen is displayed.

Figure 5-25: Sigtran Interface IDs Screen



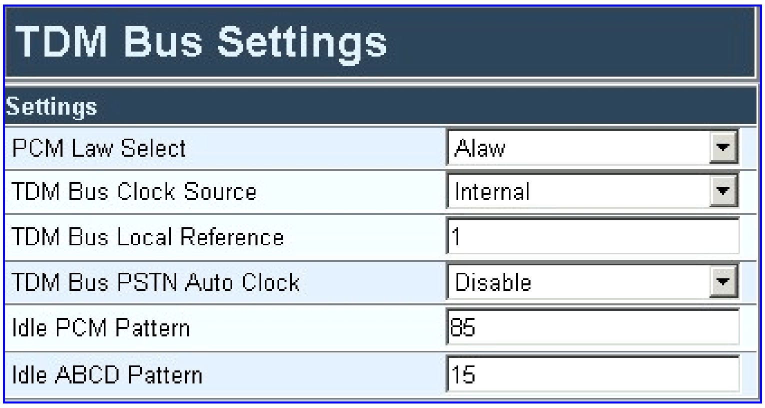
2. Configure or modify the parameters as desired (for a description of the parameters, refer to Section 12.2.4 on page 272).
3. Click **Create**.
4. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

5.6.5 Configuring the TDM Bus Settings

➢ To configure the TDM Bus Settings parameters, take these 5 steps:

1. Open the 'TDM Bus Settings' screen (**Advanced Configuration** menu > **TDM Bus Settings**); the 'TDM Bus Settings' screen is displayed.
2. Configure the TDM Bus Settings parameters.
3. Click the **Submit** button to save your changes.
4. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.
5. A device reset is required to activate the TDM Bus Settings parameters. To reset the gateway, refer to Section 5.9.3 on page 125.

Figure 5-26: TDM Bus Settings Screen



Note: Usually the 'PCM Law Select' parameter is set to A-law for E1 trunks and to µ-law for T1 trunks.

Refer to Section 10.1 on page 243 for information on configuring the 'TDM Bus Clock Source', 'TDM Bus Enable Fallback' and 'TDM Bus PSTN Auto Clock' parameters.

5.6.6 Restoring and Backing up the Gateway Configuration

The Configuration File screen enables you to restore (load a new *ini* file to the gateway) or to back up (make a copy of the VoIP gateway *ini* file and store it in a directory on your computer) the current configuration the gateway is using.

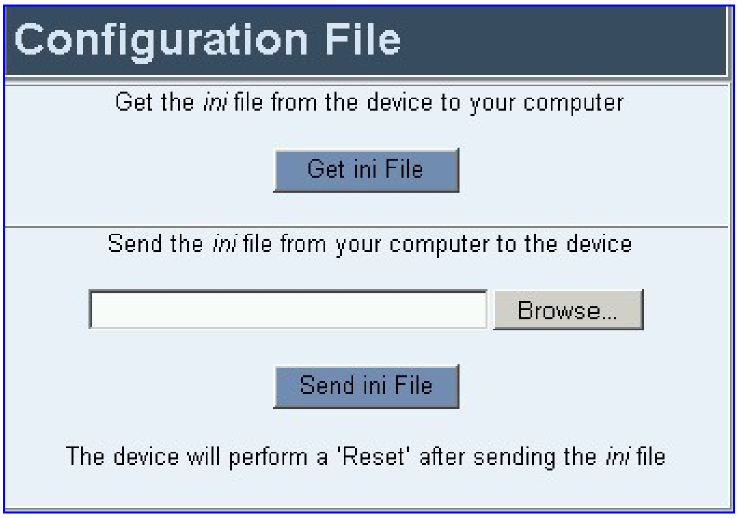
Back up your configuration if you want to protect your VoIP gateway programming. The backup *ini* file includes only those parameters that were modified and contain other than default values.

Restore your configuration if the VoIP gateway has been replaced or has lost its programming information, you can restore the VoIP gateway configuration from a previous backup or from a newly created *ini* file. To restore the VoIP gateway configuration from a previous backup you must have a backup of the VoIP gateway information stored on your computer.

➢ To restore or back up the *ini* file:

- Open the 'Configuration File' screen (**Advanced Configuration** menu > **Configuration File**); the 'Configuration File' screen is displayed.

Figure 5-27: Configuration File Screen



➢ To back up the *ini* file, take these 4 steps:

1. Click the **Get ini File** button; the 'File Download' window opens.
2. Click the **Save** button; the 'Save As' window opens.
3. Navigate to the folder where you want to save the *ini* file.
4. Click the **Save** button; the VoIP gateway copies the *ini* file into the folder you selected.

➢ To restore the *ini* file, take these 4 steps:

1. Click the **Browse** button.
2. Navigate to the folder that contains the *ini* file you want to load.
3. Click the file and click the **Open** button; the name and path of the file appear in the field beside the Browse button.
4. Click the **Send *ini* File** button, and click **OK** in the prompt; the gateway is automatically reset (from the *cmp* version stored on the flash memory).

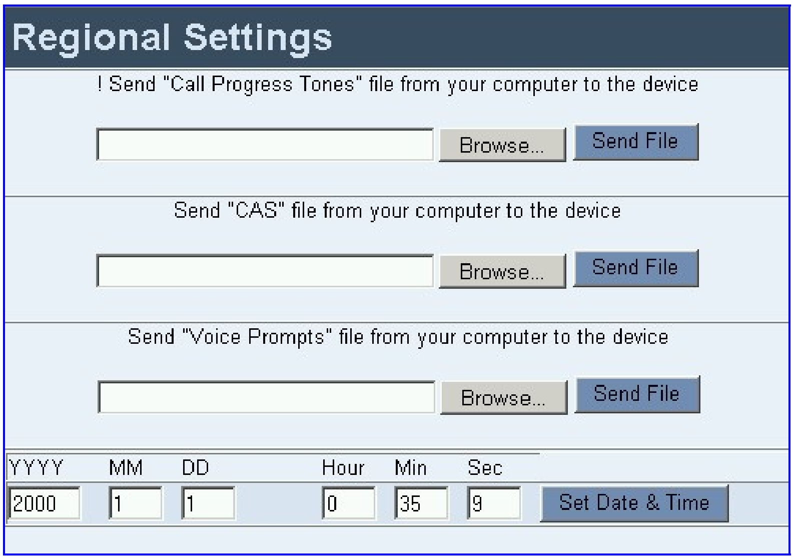
5.6.7 Regional Settings

The ‘Regional Settings’ screen enables you to set and view the gateway’s internal date and time and to load to the gateway the following configuration files: Call Progress Tones, CAS and Voice Prompts. For detailed information on the configuration files, refer to Section 16 on page 329.

➢ To configure the date and time of the gateway, take these 3 steps:

1. Open the ‘Regional Settings’ screen (**Advanced Configuration** menu > **Regional Settings**); the ‘Regional Settings' screen is displayed.

Figure 5-28: Regional Settings Screen



2. Enter the time and date where the gateway is installed.
3. Click the **Set Date & Time** button; the date and time are automatically updated.

Note: After performing a hardware reset, the date and time are returned to their defaults and should be updated.

➢ To load a configuration file to the VoIP gateway, take these 8 steps:

1. Open the ‘Regional Settings’ screen (**Advanced Configuration** menu > **Regional Settings**); the ‘Regional Settings’ screen is displayed (shown in Figure 5-28).
2. Click the **Browse** button adjacent to the file you want to load.
3. Navigate to the folder that contains the file you want to load.
4. Click the file and click the **Open** button; the name and path of the file appear in the field beside the **Browse** button.
5. Click the **Send File** button that is next to the field that contains the name of the file you want to load. An exclamation mark in the screen section indicates that the file’s loading doesn’t take effect on-the-fly (e.g., CPT file).
6. Repeat steps 2 to 5 for each file you want to load.

7. To save the loaded auxiliary files so they are available after a power fail, refer to Section 5.9.2 on page 124.
8. To reset the gateway, refer to Section 5.9.3 on page 125.

Notes:

- Saving a configuration file to flash memory may disrupt traffic on the gateway. To avoid this, perform a graceful lock (Section 5.9.1 on page 122) on all traffic on the device before saving to flash memory.
- A device reset is required to activate a loaded CPT file.

5.6.8 Security Settings

From the Security Settings you can:

- Configure the Web User Accounts (refer to Section 5.6.8.1 below).
- Configure the Web & Telnet Access List (refer to Section 5.6.8.2 on page 100).
- Configure the Firewall Settings (refer to Section 5.6.8.3 on page 101).
- Configure the Certificates (refer to Section 5.6.8.4 on page 102).
- Configure the General Security Settings.
- Configure the IPSec Table (refer to Section 13.1.3.2 on page 286).
- Configure the IKE Table (refer to Section 13.1.3.1 on page 283).

5.6.8.1 Configuring the Web User Accounts

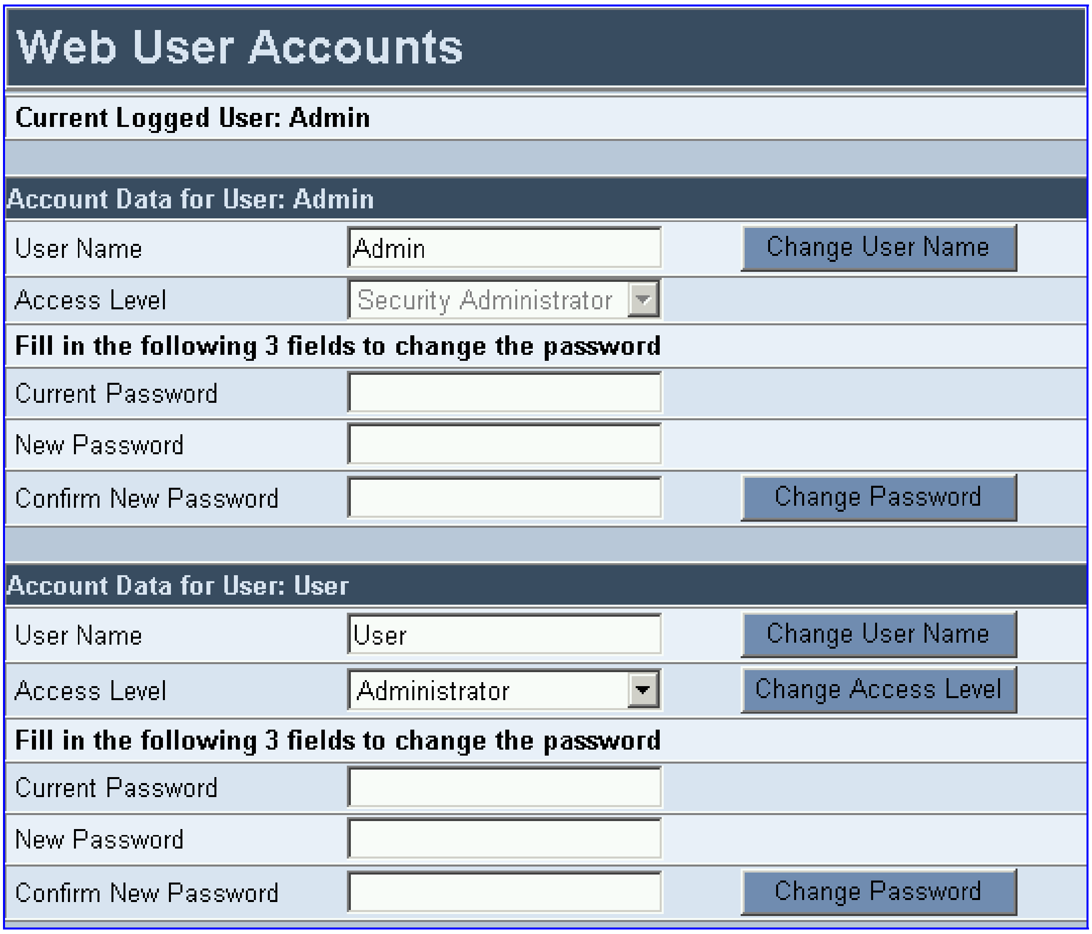
To prevent unauthorized access to the Embedded Web Server, two user accounts are available, a primary and secondary. Each account is composed of three attributes: username, password and access level. For detailed information on the user account mechanism, refer to Section 5.2.1 on page 56.

It is recommended that you change the default username and password of the account you use to access the Embedded Web Server.

➢ To change the Web User Accounts attributes, take these 4 steps:

1. Open the ‘Web User Accounts’ screen (**Advanced Configuration** menu > **Security Settings** > **Web User Accounts** option); the ‘Web User Accounts’ screen is displayed.

Figure 5-29: Web User Accounts Screen (for Users with 'Security Administrator' Privileges)



2. To change the access level of the secondary account (the access level of the primary account cannot be changed), in the 'Access Level' drop-down list, select the new access level and click the button **Change Access Level**; the new access level is applied immediately.
3. To change the username of an account, enter the new username in the field 'User Name' and click the button **Change User Name**; the new username is applied immediately and the 'Enter Network Password' screen appears (shown in Figure 5-1 on page 58). Enter the updated username in the 'Enter Network Password' screen. Note that the username can be a maximum of 19 case-sensitive characters.
4. To change the password of an account, enter the current password in the field 'Current Password', the new password in the fields 'New Password' and 'Confirm New Password' and click the button **Change Password**; the new password is applied immediately and the 'Enter Network Password' screen appears (shown in Figure 5-1 on page 58). Enter the updated password in the 'Enter Network Password' screen. Note that the password can be a maximum of 19 case-sensitive characters.

Note: A user with a 'Security Administrator' access level can change all attributes for all accounts. Users with an access level other than 'Security Administrator' can only change their own password and username.

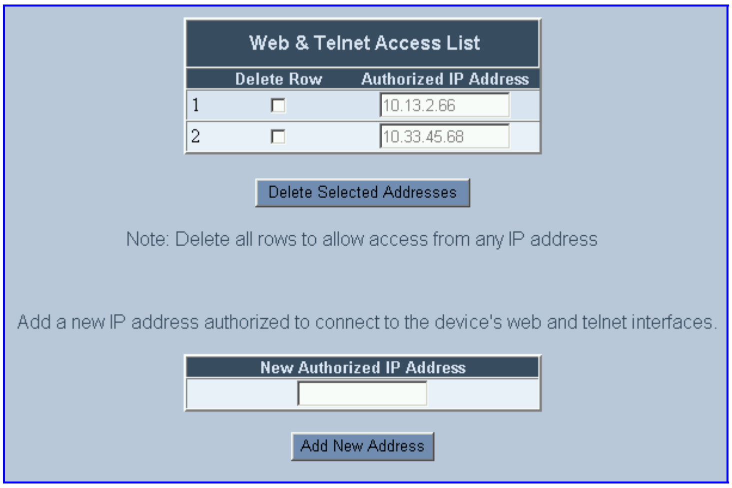
5.6.8.2 Configuring the Web and Telnet Access List

Use this screen to define up to ten IP addresses that are permitted to access the gateway's Web and Telnet interfaces. Access from an undefined IP address is denied. This security feature is inactive (the gateway can be accessed from any IP address) when the table is empty.

➢ To manage the Web & Telnet access list, take these 4 steps:

1. Open the 'Web & Telnet Access List' screen (**Advanced Configuration** menu > **Network Settings** > **Web & Telnet Access List** option); the 'Web & Telnet Access List' screen is displayed.

Figure 5-30: Web & Telnet Access List Screen



2. To add a new authorized IP address, in the 'New Authorized IP Address' field, enter the required IP address (refer to Note 1 below) and click the button **Add New Address**; the IP address you entered is added as a new entry to the Web & Telnet Access List table.
3. To delete authorized IP addresses, check the Delete Row checkbox in the rows of the IP addresses you want to delete (refer to Note 2 below) and click the button **Delete Selected Addresses**; the IP addresses are removed from the table and can no longer access the Web & Telnet interfaces.
4. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

Notes:

- The first authorized IP address you add must be your own terminal's IP address. If it isn't, further access from your terminal is denied.
- Delete your terminal's IP address from the Web & Telnet Access List last. If it is deleted before the last, access from your terminal is denied from the point of its deletion on.

5.6.8.3 Configuring the Firewall Settings

The gateway accommodates an internal Firewall, allowing the security administrator to define network traffic filtering rules. For detailed information on the internal Firewall, refer to Section 13.5 on page 298.

➢ To create a new access rule, take these 6 steps:

1. Open the ‘Firewall Settings’ screen (**Advanced Configuration** menu > **Security Settings** > **Firewall Settings** option); the ‘Firewall Settings’ screen is displayed.

Figure 5-31: Firewall Settings Screen

Firewall Settings

Selected Rule	Is Rule Active?	Source IP	Mask	Local Port Range	Protocol	Packet Size	Byte rate	Burst Bytes	Action Upon Match	Match Count
0	No	mgmt.customer.com	255.255.255.255	0-80	tcp	0	0	0	ALLOW	0
1	No	192.0.0.0	255.0.0.0	0-65535	Any	0	40000	50000	BLOCK	0
2	Yes	10.31.4.0	255.255.255.0	4000 - 9000	Any	0	0	0	Block	0
3	Yes	10.4.0.0	255.255.0.0	4000-9000	Any	0	0	0	BLOCK	0

2. In the ‘New Rule Index’ field, enter the index of the access rule that you want to add.
3. Click the **Add an Empty Rule** button; a new rule appears; alternatively, click the **Copy Selected Rule as a New Rule** button; a new rule that is an exact copy of the currently selected rule appears.
4. Configure the rule’s parameters (refer to Table 6-1 on page 130).
5. Click one of the following buttons:
   - **Apply Rule Settings** to save the new rule (the rule isn’t active).
   - **Activate Rule** to save the new rule and activate it.
   - **Delete Rule** to delete the rule.
6. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

➢ To edit a rule, take these 5 steps:

1. Click the radio button of the entry you want to edit..
2. Click the **Make Rule Editable** button; the rule’s fields can now be modified.
3. Modify the fields according to your requirements.
4. Click the **Apply Rule Settings** button to save the changes.
5. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

➢ To activate a de-activated rule, take these 2 steps:

1. Click the radio button of the entry you want to activate.
2. Click the **Activate Rule** button; the rule is active.

➢ To de-activate an activate rule, take these 2 steps:

1. Click the radio button of the entry you want to activate.
2. Click the **DeActivate Rule** button; the rule is de-activated.

➢ To delete a rule, take these 3 steps:

1. Click the radio button of the entry you want to activate.
2. Click the **Delete Rule** button; the rule is deleted.
3. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

5.6.8.4 Configuring the Certificates

Use the Certificates screen to replace the server (refer to Section 13.2.4 on page 291) and client (refer to Section 13.2.5 on page 293) certificates and to update the private key (HTTPSPkeyFileName, described in Table 6-3 on page 143).

5.6.8.5 Configuring the IPSec Table

Use the IPSec Table screen to configure the IPSec parameters. For detailed information on IPSec and IKE, refer to Section 13.1 on page 281.

5.6.8.6 Configuring the IKE Table

Use the IKE Table screen to configure the IKE parameters. For detailed information on IPSec and IKE, refer to Section 13.1 on page 281.

5.6.9 Configuring the Management Settings

From the Management Settings you can:

- Configure the Syslog Settings.
- Configure the SNMP Managers Table (refer to Section 5.6.9.1 below).
- Configure the SNMP Community Strings (refer to Section 5.6.9.2 on page 104).
- SNMP v3 Users (refer to Section 5.6.9.3 on page 105).

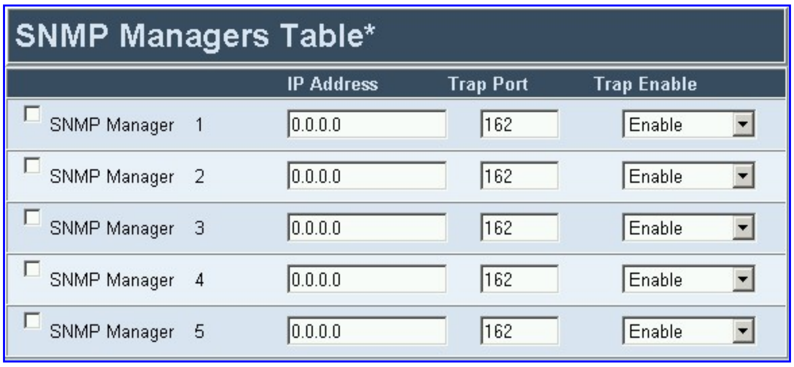
5.6.9.1 Configuring the SNMP Managers Table

The SNMP Managers table allows you to configure the attributes of up to five SNMP managers.

➢ To configure the SNMP Managers Table, take these 5 steps:

1. Access the 'Management Settings' screen (**Advanced Configuration** menu > **Management Settings**); the 'Management Settings' screen is displayed.
2. Open the SNMP Managers Table screen by clicking the arrow sign (-->) to the right of the SNMP Managers Table label; the SNMP Managers Table screen is displayed (Figure 5-32).
3. Configure the SNMP manager's parameters.
4. Click the **Submit** button to save your changes.
5. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

Figure 5-32: SNMP Managers Table Screen



Note: If you clear a checkbox and click **Submit**, all settings in the same row revert to their defaults.

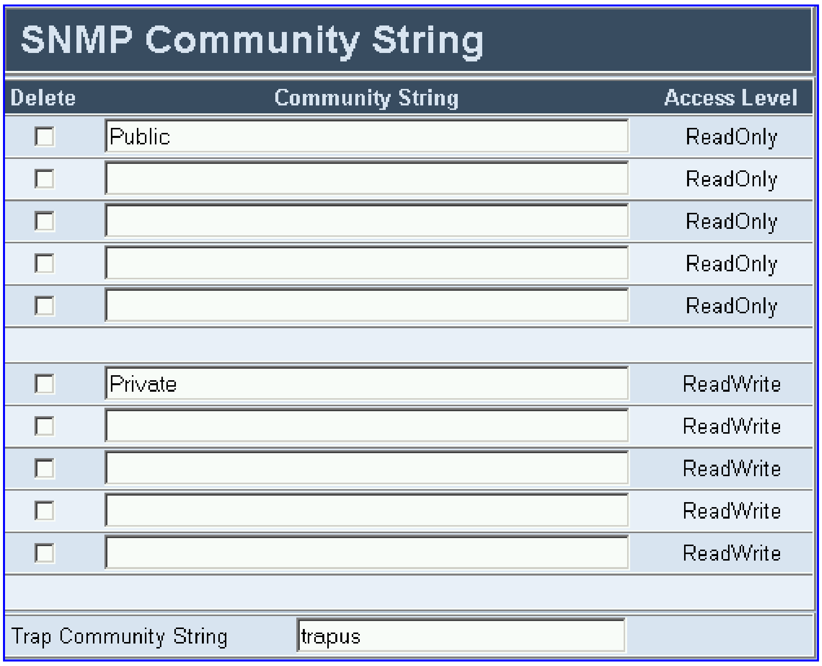
5.6.9.2 Configuring the SNMP Community Strings

Use the SNMP Community Strings table to configure up to five read-only and up to five read / write SNMP community strings, and to configure the community string that is used for sending traps. For detailed information on SNMP community strings, refer to section 15.7.1 on page 316.

➢ To configure the SNMP Community Strings, take these 5 steps:

1. Access the ‘Management Settings’ screen (**Advanced Configuration** menu > **Management Settings**); the ‘Management Settings’ screen is displayed.
2. Open the SNMP Community Strings screen by clicking the arrow sign (-->) to the right of the SNMP Community Strings label; the SNMP Community Strings screen is displayed (Figure 5-33).
3. Configure the SNMP Community Strings parameters according to Table 6-6 on page 148.
4. Click the **Submit** button to save your changes.
5. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

Figure 5-33: SNMP Community Strings Screen



Note: To delete a community string, check the **Delete** checkbox to the left of the community string you want to delete and click the button **Submit**.

5.6.9.3 Configuring SNMP v3 Users

Use the SNMP V3 Table to configure authentication and privacy for up to 10 SNMP v3 users.

➢ To configure the SNMP v3 users, take these 6 steps:

1. Access the ‘Management Settings’ screen (**Advanced Configuration** menu > **Management Settings**); the ‘Management Settings’ screen is displayed.
2. Open the 'SNMP V3 Setting' screen by clicking the arrow sign (-->) to the right of the SNMP V3 Table label; the 'SNMP V3 Setting' screen is displayed (Figure 5-34).
3. To add an SNMP v3 user, in the 'New Row Index' field, type the desired row index, and then click **Add an Empty Row**. A new row appears.
4. Configure the SNMP V3 Setting parameters according to Table 6-6 on page 148.
5. Click the **Apply Row Settings** button to save your changes.
6. To save the changes so they are available after a hardware reset or power fail, refer to Section 5.9.2 on page 124.

Figure 5-34: SNMP V3 Setting Screen

Notes:

- To delete an SNMP V3 user, select the Index radio button corresponding to the SNMP V3 user row entry that you want to delete, and then click the **Delete Row** button.
- To copy an existing SNMP V3 user configuration to a new row, select the radio button on the left of the desired SNMP V3 user row, and then click **Copy Selected Row as A New Row**. A new row appears that includes the same configuration as the selected row.
- To sort all row indexes incrementally, click **Compact Table**.

5.7 Status & Diagnostic

Use this menu to view and monitor the gateway's channels, Syslog messages, hardware / software product information, and to assess the gateway's statistics and IP connectivity information.

5.7.1 Gateway Statistics

Use the screens under Gateway Statistics to monitor real-time activity such as IP Connectivity information, call details and call statistics, including the number of call attempts, failed calls, fax calls, etc.

Note: The Gateway Statistics screens doesn't refresh automatically. To view updated information, re-access the screen you require.

5.7.1.1 IP Connectivity

The IP Connectivity screen provides you with an online read-only network diagnostic connectivity information on all destination IP addresses configured in the Tel to IP Routing table.

Notes:

- This information is available only if the parameter 'AltRoutingTel2IPEnable' (described in Table 6-10) is set to 1 (Enable) or 2 (Status Only).
- The information in columns 'Quality Status' and 'Quality Info.' (per IP address) is reset if two minutes elapse without a call to that destination.

➢ To view the IP connectivity information, take these 2 steps:

1. Set 'AltRoutingTel2IPEnable' to 1 or 2.
2. Open the 'IP Connectivity' screen (**Status & Diagnostics** menu > **Gateway Statistics** submenu > **IP Connectivity**); the 'IP Connectivity' screen is displayed (Figure 5-35).

Figure 5-35: IP Connectivity Screen

IP Connectivity

	IP Address	Host Name	Connectivity Method	Connectivity Status	Quality Status	Quality Info.	DNS Status
1	10.13.77.7	10.13.77.7	Ping	CON_OK	QOS_UNKNOWN	PL[percent]:0 DELAY [msec]:0	DNS_DISABLE
2	10.13.77.9	10.13.77.9	Ping	CON_OK	QOS_UNKNOWN	PL[percent]:0 DELAY [msec]:0	DNS_DISABLE
3	10.13.77.18	10.13.77.18	Ping	CON_FAIL	QOS_UNKNOWN	PL[percent]:0 DELAY [msec]:0	DNS_DISABLE
4	1.2.3.4	doron_pc	Ping	CON_FAIL	QOS_UNKNOWN	PL[percent]:0 DELAY [msec]:0	DNS_RESOLVED
5	10.13.2.95	xyz	Ping	CON_INIT	QOS_UNKNOWN	PL[percent]:0 DELAY [msec]:0	DNS_UNRESOLVED
6	UNUSED ENTRY	---	---	---	---	---	---
7	UNUSED ENTRY	---	---	---	---	---	---

Table 5-12: IP Connectivity Parameters

Column Name	Description
IP Address	IP address defined in the destination IP address field in the Tel to IP Routing table. or IP address that is resolved from the host name defined in the destination IP address field in the Tel to IP Routing table.
Host Name	Host name (or IP address) defined in the destination IP address field in the Tel to IP Routing table.
Connectivity Method	The method according to which the destination IP address is queried periodically (currently only by ping).
Connectivity Status	Displays the status of the IP address' connectivity according to the method in the 'Connectivity Method' field. Can be one of the following: • OK = Remote side responds to periodic connectivity queries. • Lost = Remote side didn't respond for a short period. • Fail = Remote side doesn't respond. • Init = Connectivity queries not started (e.g., IP address not resolved). • Disable = The connectivity option is disabled ('AltRoutingTel2IPMode' equals 0 or 2).
Quality Status	Determines the QoS (according to packet loss and delay) of the IP address. Can be one of the following: • Unknown = Recent quality information isn't available. • OK • Poor **Note 1:** This field is applicable only if the parameter 'AltRoutingTel2IPMode' is set to 2 or 3. **Note 2:** This field is reset if no QoS information is received for 2 minutes.
Quality Info.	Displays QoS information: delay and packet loss, calculated according to previous calls. **Note 1:** This field is applicable only if the parameter 'AltRoutingTel2IPMode' is set to 2 or 3. **Note 2:** This field is reset if no QoS information is received for 2 minutes.
DNS Status	Can be one of the following: • DNS Disable • DNS Resolved • DNS Unresolved

5.7.1.2 Call Counters

The Call Counters screens provide you with statistic information on incoming (IP→Tel) and outgoing (Tel→IP) calls. The statistic information is updated according to the release reason that is received after a call is terminated (during the same time as the end-of-call CDR message is sent). The release reason can be viewed in the Termination Reason field in the CDR message. For detailed information on each counter, refer to Table 5-13 on page 108.

You can reset this information by clicking the **Reset Counters** button.

➢ To view the IP→Tel and Tel→IP Call Counters information, take this step:

- Open the Call Counters screen you want to view (**Status & Diagnostics** menu > **Gateway Statistics** submenu); the relevant Call Counters screen is displayed. Figure 5-36 shows the 'Tel→IP Call Counters' screen.

Figure 5-36: Tel→IP Call Counters Screen

Tel to IP Calls Count	
Number of Attempted Calls	10
Number of Established Calls	5
Percentage of Successful Calls	50.000000
Number of Failed Calls due to a Busy Line	1
Number of Failed Calls due to No Answer	3
Number of Failed Calls due to No Route	0
Number of Failed Calls due to No Matched Capabilities	0
Number of Failed Calls due to Other Failures	1
Average Call Duration [sec]	15
Attempted Fax Calls Counter	0
Successful Fax Calls Counter	0

Table 5-13: Call Counters Description (continues on pages 108 to 109)

Counter	Description
Number of Attempted Calls	This counter indicates the number of attempted calls. It is composed of established and failed calls. The number of established calls is represented by the 'Number of Established Calls' counter. The number of failed calls is represented by the five failed-call counters. Only one of the established / failed call counters is incremented every time.
Number of Established Calls	This counter indicates the number of established calls. It is incremented as a result of one of the following release reasons, if the duration of the call is bigger then zero: GWAPP_REASON_NOT_RELEVANT (0) GWAPP_NORMAL_CALL_CLEAR (16) GWAPP_NORMAL_UNSPECIFIED (31) And the internal reasons: RELEASE_BECAUSE_UNKNOWN_REASON RELEASE_BECAUSE_REMOTE_CANCEL_CALL RELEASE_BECAUSE_MANUAL_DISC RELEASE_BECAUSE_SILENCE_DISC RELEASE_BECAUSE_DISCONNECT_CODE **Note:** When the duration of the call is zero, the release reason GWAPP_NORMAL_CALL_CLEAR increments the 'Number of Failed Calls due to No Answer' counter. The rest of the release reasons increment the 'Number of Failed Calls due to Other Failures' counter.

Table 5-13: Call Counters Description (continues on pages 108 to 109)

Counter	Description
Number of Failed Calls due to a Busy Line	This counter indicates the number of calls that failed as a result of a busy line. It is incremented as a result of the following release reason: GWAPP_USER_BUSY (17)
Number of Failed Calls due to No Answer	This counter indicates the number of calls that weren't answered. It is incremented as a result of one of the following release reasons: GWAPP_NO_USER_RESPONDING (18) GWAPP_NO_ANSWER_FROM_USER_ALERTED (19) And (when the call duration is zero) as a result of the following: GWAPP_NORMAL_CALL_CLEAR (16)
Number of Failed Calls due to No Route	This counter indicates the number of calls whose destinations weren't found. It is incremented as a result of one of the following release reasons: GWAPP_UNASSIGNED_NUMBER (1) GWAPP_NO_ROUTE_TO_DESTINATION (3)
Number of Failed Calls due to No Matched Capabilities	This counter indicates the number of calls that failed due to mismatched gateway capabilities. It is incremented as a result of an internal identification of capability mismatch. This mismatch is reflected to CDR via the value of the parameter 'DefaultReleaseReason' (default is GWAPP_NO_ROUTE_TO_DESTINATION (3)), or by the GWAPP_SERVICE_NOT_IMPLEMENTED_UNSPECIFIED(79) reason.
Number of Failed Calls due to Other Failures	This counter is incremented as a result of calls that fail due to reasons not covered by the other counters.
Percentage of Successful Calls	The percentage of established calls from attempted calls.
Average Call Duration [sec]	The average call duration of established calls.
Attempted Fax Calls Counter	This counter indicates the number of attempted fax calls.
Successful Fax Calls Counter	This counter indicates the number of successful fax calls.

5.7.1.3 Call Routing Status

The Call Routing Status screen provides you with information on the current routing method used by the gateway. This information includes the IP address and FQDN (if used) of the Proxy server the gateway currently operates with.

Figure 5-37: Call Routing Status Screen

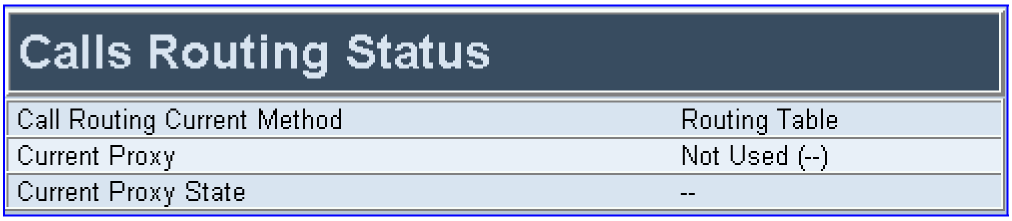


Table 5-14: Call Routing Status Parameters

Parameter	Description
Current Call-Routing Method	Proxy = Proxy server is used to route calls. Routing Table preferred to Proxy = The Tel to IP Routing table takes precedence over a Proxy for routing calls (PreferRouteTable = 1). Routing Table = The Tel to IP Routing table is used to route calls.
Current Proxy	Not Used = Proxy server isn't defined. IP address and FQDN (if exists) of the Proxy server the gateway currently operates with.
Current Proxy State	N/A = Proxy server isn't defined. OK = Communication with the Proxy server is in order. Fail = No response from any of the defined Proxies.

5.7.2 Monitoring the Gateway's Trunks & Channels

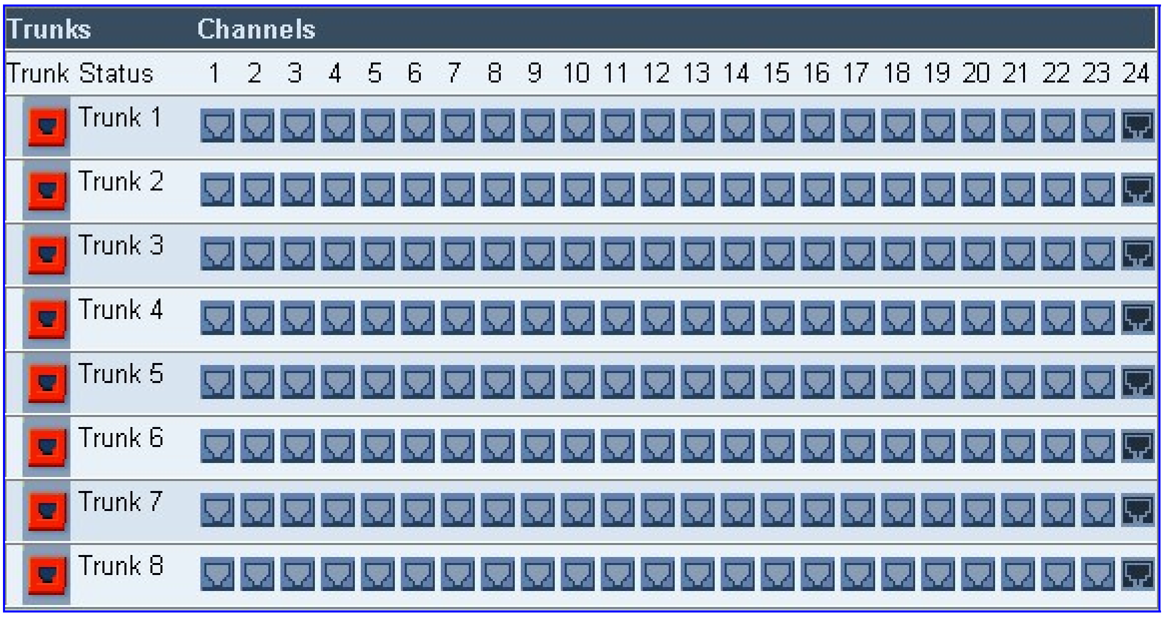
The Trunk & Channel Status screen provides real time monitoring on the current status of the gateway's trunks & channels. This screen also allows you to assign a brief description to each trunk.

The 'Trunk & Channel Status' screen is easily accessed using the Home icon located above the main menu bar.

➢ To monitor the status of the trunks and B-channels, take this step:

- Open the 'Trunk & Channel Status' screen by clicking the Home icon; the 'Trunk & Channel Status' screen is displayed.

Figure 5-38: Trunk & Channel Status Screen



The number of trunks and channels that appear on the screen depends on the system configuration. The example above depicts a system with 8 T1 spans.

The trunk and channel status indicators appear colored. Figure 5-39 shows the possible indicators and their descriptions.

Figure 5-39: Trunk and Channel Status Color Indicator Keys

➢ To monitor the details of a specific channel, take these 2 steps:

1. Click the numbered icon of the specific channel whose detailed status you need to view; the channel-specific Channel Status screen appears, shown in Figure 5-40.
2. Click the submenu links to view a specific channel's parameter settings.

Figure 5-40: Channel Status Details Screen

SIP Channel Status

Static Information		
Endpoint Status :	ACTIVE	
Assigned Phone Number :	100	
Trunk Group :	default (0)	
MWI Information :	--	
Associated Calls Information		
Call ID :	265821508dMlu@10.8.58.1	--
Call Originator :	TEL	--
Source Tel Number :	100	--
Destination Tel Number :	200	--
Redirect Calling Number :		--
Remote Signaling IP :	10.8.58.2	--
Remote RTP (IP:Port) :	10.8.58.2: 4000	--
Call Establishment Duration :	2	--
Call Duration :	17	--
Call State :	SESSION	--
Fax State :	n/a	--
Coder + PTime :	g7231:30	--
Call Type :	Voice	--
Call Establishment Method :	Normal	--
DTMF Selected Method for Tx/Rx :	DTMF_NOT_SUPPORTED	--

➢ To add a port description, take these 3 steps:

1. Open the 'Channel Status' screen by clicking the Home icon.
2. Click the desired trunk icon, and then from the shortcut menu, choose **Update Port Info**; a text box appears.
3. Type a brief description for this trunk, and then click **Apply Port Info**.

5.7.3 Activating the Internal Syslog Viewer

The Message Log screen displays Syslog debug messages sent by the gateway.

Note that it is not recommended to keep a 'Message Log' session open for a prolonged period (refer to the Note below). For prolong debugging use an external Syslog server, refer to Section 14.2 on page 304.

Refer to the Debug Level parameter 'GwDebugLevel' (described in Table 6-2) to determine the Syslog logging level.

➢ To activate the Message Log, take these 4 steps:

1. In the **General Parameters** screen under **Advanced Parameters** submenu (accessed from the **Protocol Management** menu), set the parameter 'Debug Level' to 5. This parameter determines the Syslog logging level, in the range 0 to 5, where 5 is the highest level.
2. Open the 'Message Log' screen (**Status & Diagnostics** menu > **Message Log**); the 'Message Log' screen is displayed and the Log is activated.

Figure 5-41: Message Log Screen

```
Log is Activated

12d:6h:56m:26s (      lgr_flow)(460       )  ---- Incoming SIP Message from 10.8.58.1:5060 ----

12d:6h:56m:26s INVITE sip:200@10.8.58.4;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.8.58.1;branch=z9hG4bKackpUGBoT
Max-Forwards: 70
From: <sip:100@10.8.58.1>;tag=1c910315947
To: <sip:200@10.8.58.4;user=phone>
Call-ID: 1254421147LEqU@10.8.58.1
CSeq: 1 INVITE
Contact: <sip:100@10.8.58.1>
Supported: em,timer,replaces,path
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE
User-Agent: Audiocodes-Sip-Gateway-MP-104 FXS/v.4.40.123.223
Content-Type: application/sdp
Content-Length: 161
```

3. Select the messages, copy them and paste them into a text editor such as Notepad. Send this *txt* file to our Technical Support for diagnosis and troubleshooting.
4. To clear the screen of messages, click on the submenu **Message Log**; the screen is cleared and new messages begin appearing.

Tip: Do not keep the 'Message Log' screen minimized for a prolonged period as a prolonged session may cause the gateway to overload. As long as the screen is open (even if minimized), a session is in progress and messages are sent. Closing the screen (and accessing another) stops the messages and terminates the session.

5.7.4 Device Information

The Device Information screen displays specific hardware and software product information. This information can help you to expedite any troubleshooting process. Capture the screen and email it to ‘our’ Technical Support personnel to ensure quick diagnosis and effective corrective action. From this screen you can also view and remove any loaded files used by the gateway (stored in the RAM).

➢ To access the Device Information screen:

- Open the ‘Device Information’ screen (**Status & Diagnostics** menu > **Device Information**); the ‘Device Information’ screen is displayed.

Figure 5-42: Device Information Screen

Device Information		
General		
MAC Address:	00908f086444	
Serial Number:	549956	
Board Type:	33	
Device Up Time:	3d:23h:47m:24s:95th	
Device Administrative State:	Unlocked	
Device Operational State:	Enabled	
Flash Size [bytes]:	8388608	
RAM Size [bytes]:	134217728	
CPU Speed [MHz]:	200	
Versions		
Version ID:	5.00A.012.009	
DSP Type:	2	
DSP Software Version:	20912	
DSP Software Name:	620AE3	
Flash Version:	192	
Module FirmWare:	0x34	
Loaded Files		
Last Loaded Voice Prompt File Name:	vp_zvi.dat	Delete
Loaded Call Progress Tones:	Default Progress Tones	
Loaded Coder Table :	Default CODERTABLE	

➢ To delete any of the loaded files, take these 4 steps:

1. Click the **Delete** button to the right of the files you want to delete. Deleting a file takes effect only after the gateway is reset.
2. Click the **Maintenance** button on the main menu bar; the 'Maintenance Actions' screen is displayed.
3. In the 'Burn to FLASH' field, select 'Yes'.
4. Click the **Reset** button. The IPmedia 2000 is reset and the files you chose to delete are discarded.

5.7.5 Viewing the Ethernet Port Information

The Ethernet Port Information screen provides read-only information on the Ethernet connection used by the gateway. The Ethernet Port Information parameters are displayed in Table 5-15. For detailed information on the Ethernet redundancy scheme, refer to Section 9.2 on page 229. For detailed information on the Ethernet interface configuration, refer to Section 9.1 on page 229.

➢ To view the Ethernet Port Information parameters:

- Open the ‘Ethernet Port Information’ screen (**Advanced Configuration** menu > **Network Settings** > **Ethernet Port Information** option); the ‘Ethernet Port Information’ screen is displayed.

Figure 5-43: Ethernet Port Information Screen

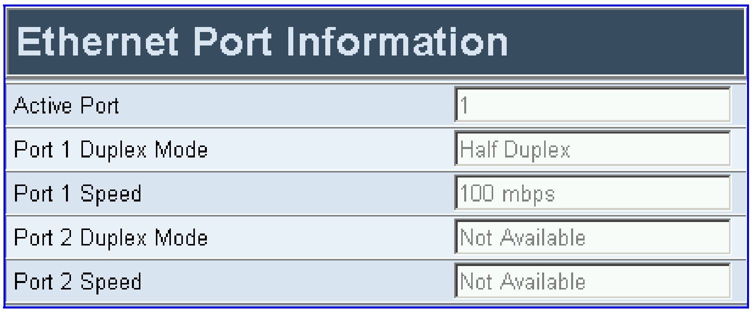


Table 5-15: Ethernet Port Information Parameters

Parameter	Description
Active Port	Shows the active Ethernet port (1 or 2).
Port 1 Duplex Mode	Shows the Duplex mode Ethernet port 1 is using (Half Duplex or Full Duplex).
Port 1 Speed	Shows the speed, in Mbps, that Ethernet port 1 is using (10 or 100 Mbps).
Port 2 Duplex Mode	Shows the Duplex mode Ethernet port 2 is using (Half Duplex or Full Duplex).
Port 2 Speed	Shows the speed, in Mbps, that Ethernet port 2 is using (10 or 100 Mbps).

5.8 Software Update Menu

The ‘Software Update’ menu enables users to upgrade the gateway's software by loading a new *cmp* file along with the *ini* and a suite of auxiliary files, or to update the existing auxiliary files.

The ‘Software Update’ menu comprises two submenus:

- Software Upgrade Wizard (refer to Section 5.8.1 below).
- Auxiliary Files (refer to Section 5.8.2 on page 119).

> **Note:** When upgrading the gateway's software you *must* load the new *cmp* file with all other related configuration files

5.8.1 Software Upgrade Wizard

The Software Upgrade Wizard guides users through the process of software upgrade: selecting files and loading them to the gateway. The wizard also enables users to upgrade software while maintaining the existing configuration. Using the wizard obligates users to load and burn a *cmp* file. Users can choose to also use the wizard to load the *ini* and auxiliary files (e.g., Call Progress Tones*)* but this option cannot be pursued without loading the *cmp* file. For the *ini* and each auxiliary file type, users can choose to reload an existing file, load a new file or not load a file at all.

The Software Upgrade Wizard allows you to load the following files:

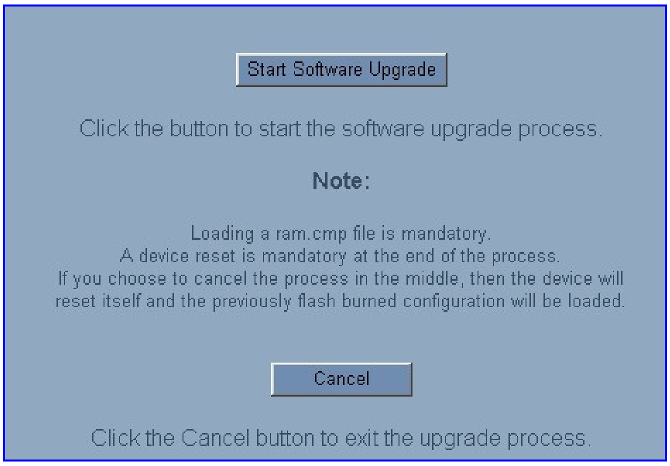
- *cmp* (mandatory)
- *ini*
- Auxiliary files:
  - CPT (Call Progress Tone)
  - VP (Voice Prompts)
  - PRT (Prerecorded Tones)
  - CAS
  - USRINF (User Info)

> **Warning 1:** The Software Upgrade Wizard requires the gateway to be reset at the end of the process, disrupting any of its traffic. To avoid disruption, disable all traffic by performing a graceful lock (refer to Section 5.9.1 on page 122) on the gateway before initiating the Wizard..
>
> **Warning 2:** Verify, prior to clicking the **Start Software Upgrade** button that no traffic is running on the device. After clicking this button a device reset is mandatory. Even if you choose to cancel the process in the middle, the device resets itself and the previous configuration burned to flash is reloaded.

➢ To use the Software Upgrade Wizard, take these 10 steps:

1. Stop all traffic on the gateway (refer to the note above).
2. Open the 'Software Upgrade Wizard' (**Software Update** menu > **Software Upgrade Wizard**); the 'Start Software Upgrade' screen appears.

Figure 5-44: Start Software Upgrade Screen

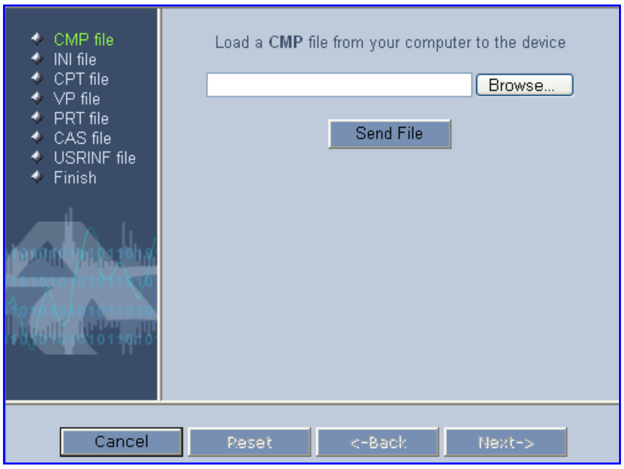


Note:	At this point, the process can be canceled with no consequence to the gateway (click the **Cancel** button). If you continue the process (by clicking the **Start Software Upgrade** button, the process must be followed through and completed with a gateway reset at the end. If you click the **Cancel** button in any of the subsequent screens, the gateway is automatically reset with the configuration that was previously burned in flash memory.

3. Click the **Start Software Upgrade** button; the 'Load a *cmp* file' screen appears (Figure 5-45).

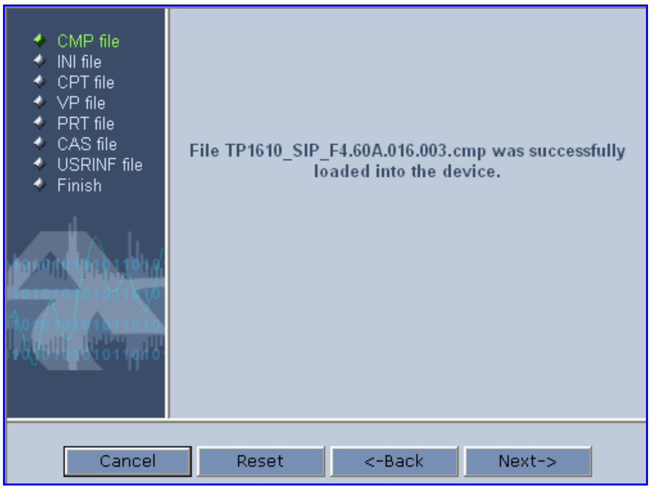
Note:	When in the Wizard process, the rest of the Web application is unavailable and the background Web screen is disabled. After the process is completed, access to the full Web application is restored.

Figure 5-45: Load a *cmp* File Screen



4. Click the **Browse** button, navigate to the *cmp* file and click the button **Send File**; the *cmp* file is loaded to the gateway and you're notified as to a successful loading (refer to Figure 5-46).

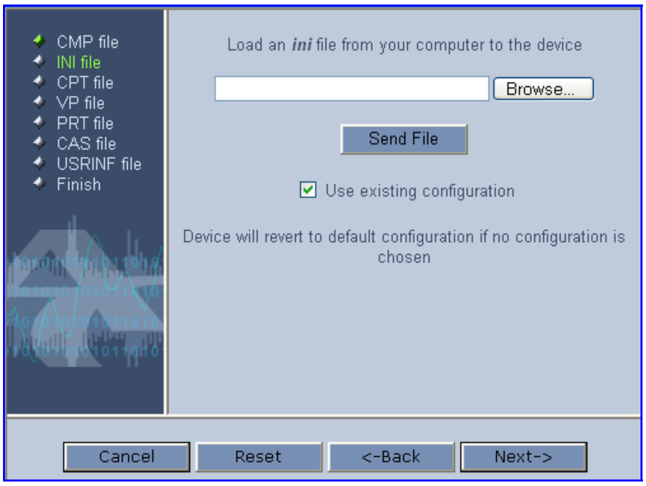
Figure 5-46: *cmp* File Successfully Loaded onto the Gateway Notification



5. Note that the four action buttons (**Cancel**, **Reset**, **Back**, and **Next**) are now activated (following *cmp* file loading).
   You can now choose to either:
   - Click **Reset**; the gateway resets, utilizing the new *cmp* you loaded and utilizing the current configuration files.
   - Click **Cancel**; the gateway resets utilizing the *cmp, ini* and all other configuration files that were previously stored in flash memory. Note that these are NOT the files you loaded in the previous Wizard steps.
   - Click **Back**; the 'Load a *cmp* File' screen is reverted to; refer to Figure 5-45.
   - Click **Next**; the 'Load an *ini* File' screen opens; refer to Figure 5-47. Loading a new *ini* file or any other auxiliary file listed in the Wizard is optional.

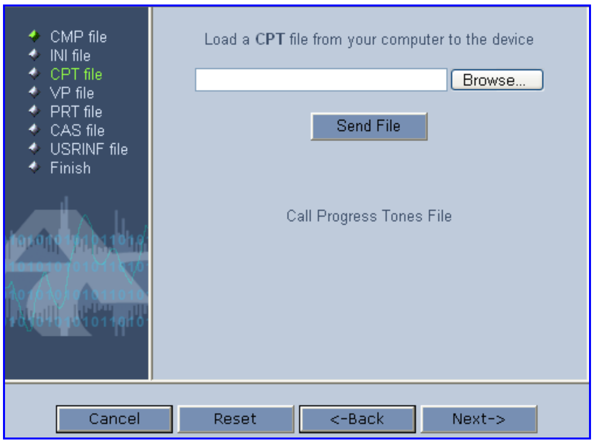
Note that as you progress, the file type list on the left indicates which file type loading is in process by illuminating green (until 'FINISH').

Figure 5-47: Load an *ini* File Screen



6. In the 'Load an *ini* File' screen, you can now choose to either:
   - Click **Browse** and navigate to the *ini* file; the check box 'Use existing configuration', by default checked, becomes unchecked. Click **Send File**; the *ini* file is loaded to the gateway and you're notified as to a successful loading.
   - Ignore the **Browse** button (its field remains undefined and the check box 'Use existing configuration' remains checked by default).
   - Ignore the **Browse** button and uncheck the 'Use existing configuration' check box; no *ini* file is loaded, the gateway uses its factory-preconfigured values.
7. You can now choose to either:
   - Click **Cancel**; the gateway resets utilizing the *cmp, ini* and all other configuration files that were previously stored in flash memory. Note that these are NOT the files you loaded in the previous Wizard steps.
   - Click **Reset**; the gateway resets, utilizing the new *cmp* and *ini* file you loaded up to now as well as utilizing the other configuration files.
   - Click **Back**; the 'Load a *cmp* file' screen is reverted to; refer to Figure 5-45.
   - Click **Next**; the 'Load a CPT File' screen opens (refer to Figure 5-48); Loading a new CPT file or any other auxiliary file listed in the Wizard is optional.

Figure 5-48: Load a CPT File Screen



8. Follow the same procedure you followed when loading the *ini* file (refer to Step 6). The same procedure applies to the 'Load a coefficient file' screen.
9. In the 'Finish' screen (refer to Figure 5-49), the **Next** button is disabled. Complete the upgrade process by clicking **Reset** or **Cancel**.
   - Click **Reset**, the gateway 'burns' the newly loaded files to flash memory. The 'Burning files to flash memory' screen appears. Wait for the 'burn' to finish. When it finishes, the 'End Process' screen appears displaying the burned configuration files (refer to Figure 5-50).
   - Click **Cancel**, the gateway resets, utilizing the files previously stored in flash memory. (Note that these are NOT the files you loaded in the previous Wizard steps).

Figure 5-49: Finish Screen

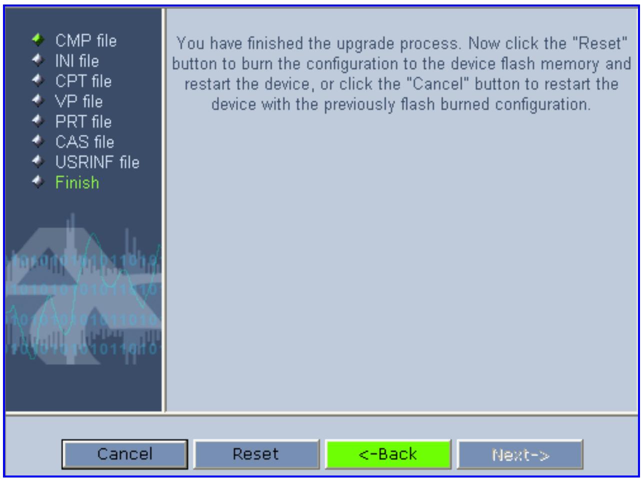
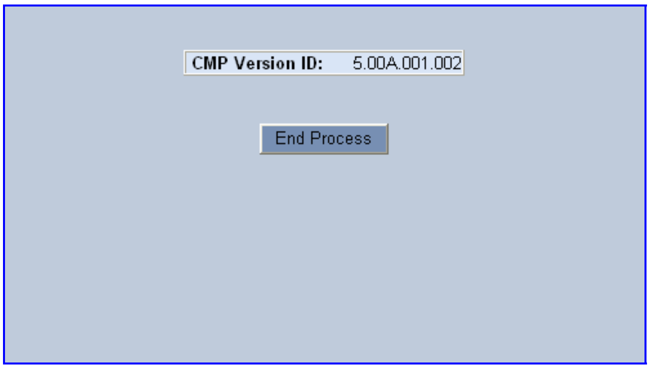


Figure 5-50: End Process Screen



10. Click the **End Process** button; the 'Quick Setup' screen appears and the full Web application is reactivated.

5.8.2 Auxiliary Files

The 'Auxiliary Files' screen enables you to load to the gateway the following files: CAS, Call Progress Tones, Voice Prompts, Prerecorded Tones (PRT) and User Information. For detailed information on these files, refer to Section 16 on page 329. For information on deleting these files from the gateway, refer to Section 5.7.4 on page 113.

Table 5-16 presents a brief description of each auxiliary file.

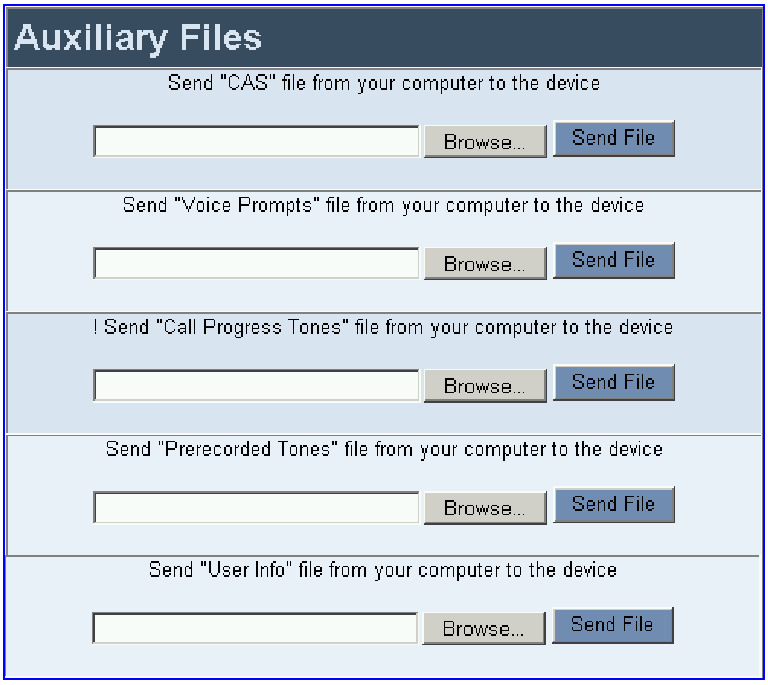
Table 5-16: Auxiliary Files Descriptions

File Type	Description
CAS	Up to 8 different CAS files containing specific CAS protocol definitions. These files are provided to support various types of CAS signaling.
Voice Prompts	The voice announcement file contains a set of Voice Prompts to be played by the gateway during operation.
Call Progress Tones	This is a region-specific, telephone exchange-dependent file that contains the Call Progress Tones levels and frequencies that the VoIP gateway uses. The default CPT file is: U.S.A.
Prerecorded Tones	The *dat* PRT file enhances the gateway's capabilities of playing a wide range of telephone exchange tones that cannot be defined in the Call Progress Tones file.
User Information	The User Information file maps PBX extensions to IP numbers. This file can be used to represent PBX extensions as IP phones in the global 'IP world'.

➢ To load an auxiliary file to the gateway, take these 8 steps:

1. Open the 'Auxiliary Files' screen (**Software Upgrade** menu > **Load Auxiliary Files**); the 'Auxiliary Files' screen is displayed.

Figure 5-51: Auxiliary Files Screen



2. Click the **Browse** button that is in the field for the type of file you want to load.
3. Navigate to the folder that contains the file you want to load.
4. Click the file and click the **Open** button; the name and path of the file appear in the field beside the **Browse** button.
5. Click the **Send File** button that is next to the field that contains the name of the file you want to load. An exclamation mark in the screen section indicates that the file's loading doesn't take effect on-the-fly (e.g., CPT file).
6. Repeat steps 2 to 5 for each file you want to load.

Notes:

- Saving an auxiliary file to flash memory may disrupt traffic on the gateway. To avoid this, disable all traffic on the device before saving to flash by performing a graceful lock (refer to Section 5.9.1 on page 122).
- A device reset is required to activate a loaded CPT file, and may be required for the activation of certain *ini* file parameters.

7. To save the loaded auxiliary files so they are available after a power fail, refer to Section 5.9.2 on page 124.
8. To reset the gateway, refer to Section 5.9.3 on page 125.

5.8.3 Updating the Software Upgrade Key

The AudioCodes gateways are supplied with a Software Upgrade Key already pre-configured for each of its TrunkPack Modules (TPM).

Users can later upgrade their gateway features, capabilities and quantity of available resources by specifying what upgrades they require, and purchasing a new key to match their specification.

The Software Upgrade Key is sent as a string in a text file, to be loaded onto the gateway. Stored in the device's non-volatile flash memory, the string defines the features and capabilities allowed by the specific key purchased by the user. The device allows users to utilize *only these* features and capabilities. A new key overwrites a previously installed key.

For detailed information on the Software Upgrade Key, refer to Section 11.7 on page 262.

5.9 Maintenance

The Maintenance menu is used for the following operations:

- Locking and unlocking the gateway (refer to Section 5.9.1 on page 122)
- Saving the gateway's configuration (refer to Section 5.9.2 on page 124)
- Resetting the gateway (refer to Section 5.9.3 on page 125)

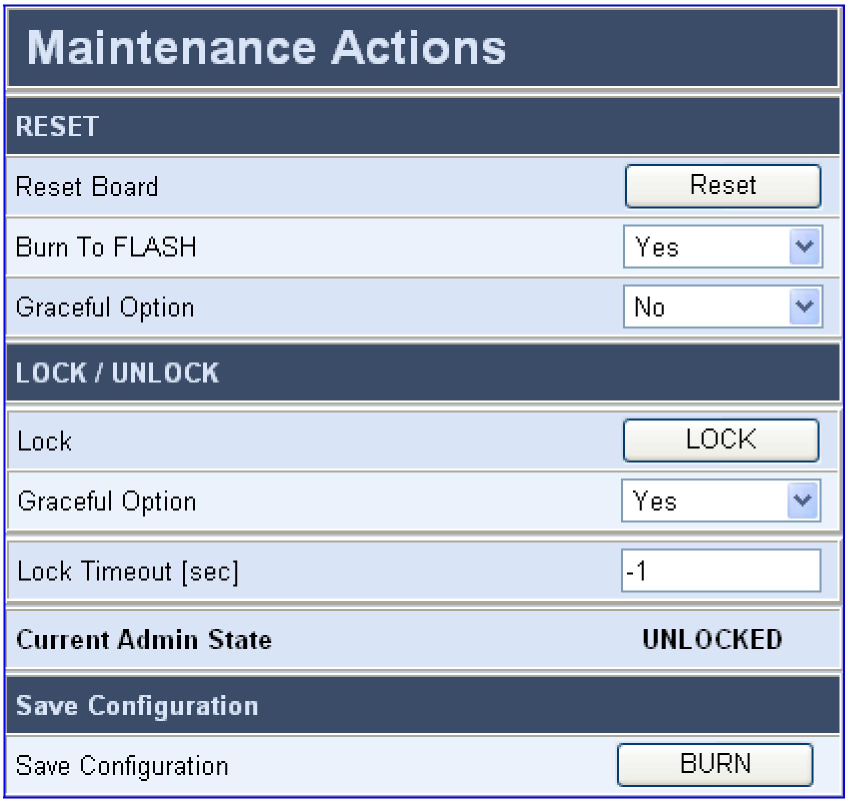
5.9.1 Locking and Unlocking the Gateway

The Lock and Unlock options allow you to lock the IPmedia 2000 so that it does not accept any new incoming calls. This is beneficial when, for example, you are uploading new software files to the gateway and you don't want any traffic to interfere with the process.

➢ To lock the IPmedia 2000, take these 4 steps:

1. Open the 'Maintenance Actions' screen (**Maintenance** menu); the 'Maintenance Actions' screen is displayed.

Figure 5-52: Maintenance Actions Screen



2. Under the LOCK / UNLOCK group, from the 'Graceful Option' drop-down list, select one of the following options:
   - 'Yes': The gateway is 'locked' only after the user-defined time in the 'Lock Timeout' field (refer to Step 3) expires or no more active traffic exists (the earliest thereof). In addition, no new traffic is accepted.
   - 'No': The gateway is "locked" regardless of traffic. Any existing traffic is terminated immediately.

3. In the 'Lock Timeout' field (relevant only if the 'Graceful Option' in the previous step is set to 'Yes'), enter the time (in seconds) after which the gateway locks. Note that if no traffic exists and the time has not expired, the gateway locks.
4. Click the **LOCK** button. If 'Graceful Option' is set to 'Yes', the lock is delayed and a screen displaying the number of remaining calls and time is displayed. Otherwise, the lock process begins immediately. The Current Admin State displays the current state: LOCKED or UNLOCKED.

➢ To unlock the IPmedia 2000, take these 2 steps:

1. Access the 'Maintenance Actions' screen as described above in the previous procedure.
2. Click the **UNLOCK** button. Unlock starts immediately and the gateway is ready for new incoming calls.

5.9.2 Saving Configuration

The 'Maintenance Actions' screen enables you to save the current parameter configuration and the loaded auxiliary files to the *non-volatile* memory (i.e., flash) so they are available after a hardware reset (or power fail). Parameters that are only saved to the *volatile* memory (RAM) revert to their previous settings after hardware reset.

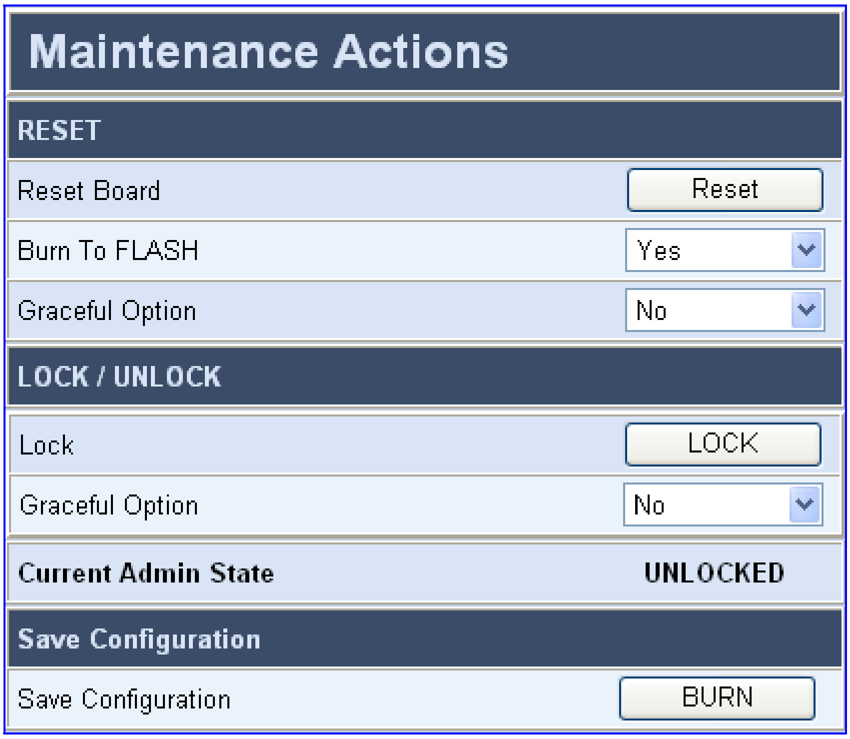
Notes:

- Saving changes to the *non-volatile* memory may disrupt traffic on the gateway. To avoid this, perform this during low-traffic periods or disable all traffic before saving, by performing a graceful lock (refer to Section 5.9.1 on page 122(.
- In the Web interface, parameters prefixed with an exclamation mark ('!') are saved to the non-volatile memory only after a device reset.
- When performing a software reset using the Web (refer to Section 5.9.3 on page 125) or SNMP, you can choose to save the changes to the *non-volatile* memory.

➢ To save configuration changes to the *non-volatile memory*, take these 2 steps:

1. Open the 'Maintenance Actions' screen (**Maintenance** menu); the 'Maintenance Actions' screen is displayed.

Figure 5-53: Maintenance Actions Screen



2. Click the **BURN** button; a confirmation message appears when the save is completed successfully.

5.9.3 Resetting the IPmedia 2000

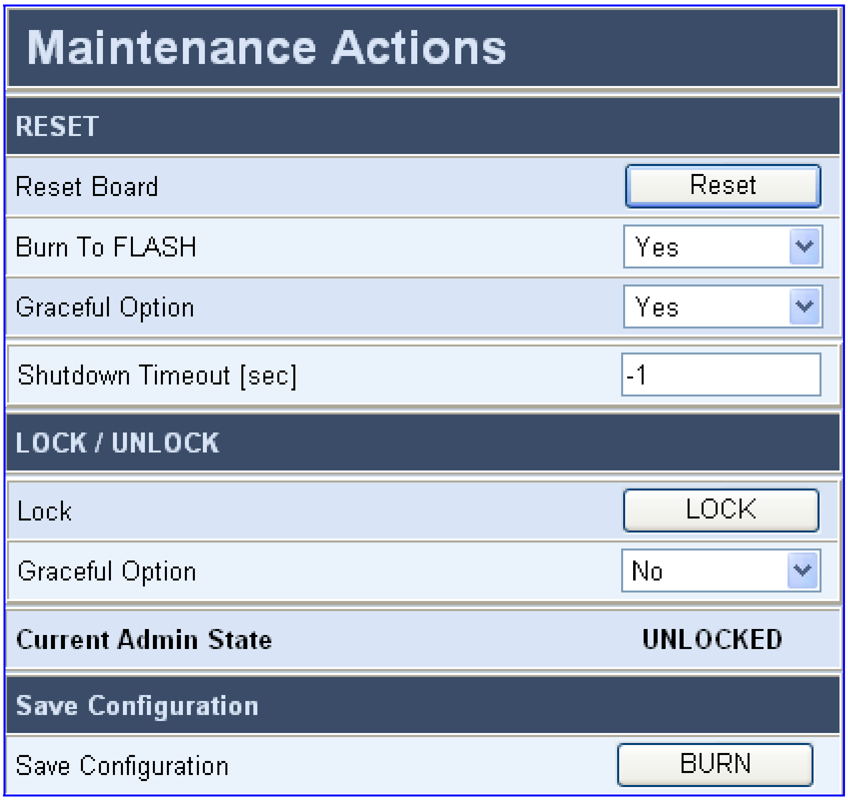
The 'Maintenance Actions' screen enables you to remotely reset the gateway. Before you reset the gateway, you can choose the following options:

- Save the gateway's current configuration to the flash memory (non-volatile) before reset.
- Perform a graceful shutdown. Reset starts only after a user-defined time expires or no more active traffic exists (the earliest thereof).

➢ To reset the IPmedia 2000, take these 5 steps:

1. Open the ‘Maintenance Actions' screen (**Maintenance** menu); the ‘Maintenance Actions' screen is displayed.

Figure 5-54: Maintenance Actions Screen



2. Under the RESET group, from the 'Burn To FLASH' drop-down list, select one of the following options:
   - 'Yes': The gateway's current configuration is burned (i.e., saved) to the flash memory prior to reset (default).
   - 'No': Resets the device without burning (i.e., saving) the current configuration to flash (discards all unsaved modifications to the configuration).
3. Under the RESET group, from the 'Graceful Option' drop-down list, select one of the following options:
   - 'Yes': Reset starts only after the user-defined time in the Shutdown Timeout field (refer to Step 4) expires or no more active traffic exists (the earliest thereof). In addition, no new traffic is accepted.
   - 'No': Reset starts regardless of traffic and any existing traffic is terminated at once.

4. In the 'Shutdown Timeout' field (relevant only if the 'Graceful Option' in the previous step is set to 'Yes'), enter the time after which the gateway resets. Note that if no traffic exists and the time has not expired, the gateway resets.
5. Click the **RESET** button. If 'Graceful Option' is set to 'Yes', the reset is delayed and a screen displaying the number of remaining calls and time is displayed.

When the device resets, a message is displayed informing of the waiting period.

Note: When a gatekeeper is used, the gateway issues an Unregister request before it is reset (either from the Embedded Web Server, SNMP or BootP).

5.10 Logging Off the Embedded Web Server

The **Log Off** button enables you to log off the Embedded Web Server and to re-access it with a different account. For detailed information on the Web User Accounts, refer to Section 5.2.1 on page 56.

➢ To log off the Embedded Web Server, take these 2 steps:

1. Click the **Log Off** button on the main menu bar; the Log Off prompt screen is displayed.

Figure 5-55: Log off Prompt

2. Click **OK**; the Web session is logged off.

6 Gateway's *ini* File Configuration

As an alternative to configuring the VoIP gateway using the Web Interface (refer to Chapter 5 on page 55), it can be configured by loading the *ini* file containing Customer-configured parameters.

The *ini* file is loaded via the BootP/TFTP utility (refer to Appendix D on page 353) or via any standard TFTP server. It can also be loaded through the Web Interface (refer to Section 5.6.6 on page 96).

The *ini* file configuration parameters are stored in the gateway's non-volatile memory after the file is loaded. When a parameter is missing from the *ini* file, a default value is assigned to that parameter (according to the *cmp* file loaded on the gateway) and stored in the non-volatile memory (thereby overriding the value previously defined for that parameter). Therefore, to restore the default configuration parameters, use the *ini* file without any valid parameters or with a semicolon (;) preceding all lines in the file.

Some of the gateway's parameters are configurable through the *ini* file only (and not via the Web). These parameters usually determine a low-level functionality and are seldom changed for a specific application.

6.1 Secured *ini* File

The *ini* file contains sensitive information that is required for the functioning of the gateway. It is loaded to, or retrieved from, the device via TFTP or HTTP. These protocols are unsecured and vulnerable to potential hackers. Therefore an encoded *ini* file significantly reduces these threats.

You can choose to load an encoded *ini* file to the gateway. When you load an encoded *ini* file, the retrieved *ini* file is also encoded. Use the ‘TrunkPack Downloadable Conversion Utility’ to encode or decode the *ini* file before you load it to, or retrieve it from, the device. Note that the encoded *ini* file’s loading procedure is identical to the regular *ini* file’s loading procedure. For information on encoding / decoding an *ini* file, refer to Section G.1.4 on page 374.

6.2 Modifying an *ini* File

➢ To modify the *ini* file, take these 3 steps:

1. Get the *ini* file from the gateway using the Embedded Web Server (refer to Section 5.6.6 on page 96).
2. Open the file (the file opens in Notepad or a Customer-defined text file editor) and modify the *ini* file parameters according to your requirements. Save and close the file.
3. Load the modified *ini* file to the gateway (using either the BootP/TFTP utility or the Embedded Web Server).

This method preserves the programming that already exists in the device, including special default values that were preconfigured when the unit was manufactured.

Tip: Before loading the *ini* file to the gateway, verify that the extension of the *ini* file saved on your PC is correct. On your PC, verify that the check box ‘Hide file extension for known file types’ (My computer > Tools > Folder Options > View) is unchecked. Then, confirm that the *ini* file name extension is xxx.ini and NOT erroneously xxx.ini.ini or xxx~.ini.

6.3 The *ini* File Content

The *ini* file contains the following SIP gateway information:

- Networking parameters shown in Table 6-1 on page 130.
- System parameters shown in Table 6-2 on page 138.
- Web and Telnet parameters shown in Table 6-3 on page 143.
- Security parameters shown in Table 6-4 on page 145.
- RADIUS parameters shown in Table 6-5 on page 147.
- SNMP parameters shown in Table 6-6 on page 148.
- SIP Configuration parameters shown in Table 6-7 on page 150.
- Voice Mail parameters shown in Table 6-8 on page 170.
- ISDN and CAS Interworking-Related Parameters shown in Table 6-9 on page 172.
- Number Manipulation and Routing parameters shown in Table 6-10 on page 180.
- E1/T1 Configuration Parameters shown in Table 6-11 on page 189.
- Channel Parameters shown in Table 6-12 on page 196.
- Configuration Files parameters shown in Section Table 6-13 on page 201.

Note: In Table 6-1 through Table 6-13, parameters in brackets are the format in the Embedded Web Server.

6.4 The *ini* File Structure

The *ini* file can contain any number of parameters. The parameters are divided into groups by their functionality. The general form of the *ini* file is shown in Figure 6-1 below.

Figure 6-1: *ini* File Structure

```
[Sub Section Name]

Parameter Name = Parameter Value
Parameter Name = Parameter Value

; REMARK

[Sub Section Name]
```

6.4.1 The *ini* File Structure Rules

- The *ini* file name mustn't include hyphens or spaces, use underscore instead.
- Lines beginning with a semi-colon ';' (as the first character) are ignored.
- A Carriage Return must be the final character of each line.
- The number of spaces before and after '=' is not relevant.
- If there is a syntax error in the parameter name, the value is ignored.
- Syntax errors in the parameter value field can cause unexpected errors (because parameters may be set to the wrong values).
- Sub-section names are optional.
- String parameters, representing file names, for example CallProgressTonesFileName, must be placed between two inverted commas ('…').

- The parameter name is NOT case-sensitive; the parameter value is NOT case-sensitive *except for coder names*.
- The *ini* file should be ended with one or more carriage returns.

6.5 The *ini* File Example

Figure 6-2 shows an example of an *ini* file for the VoIP gateway.

Figure 6-2: SIP *ini* File Example

```
PCMLawSelect = 1
ProtocolType = 1
TerminationSide = 0
FramingMethod = 0
LineCode = 2
TDMBusClockSource = 4
ClockMaster = 0

;Channel Params
DJBufMinDelay = 75
RTPRedundancyDepth = 1

IsProxyUsed = 1
ProxyIP = 192.168.122.179

CoderName = g7231,90

;List of serial B-channel numbers

TrunkGroup_1 = 0/1-24,1000
TrunkGroup_2 = 1/1-24,2000
TrunkGroup_3 = 2/1-24,3000
TrunkGroup_4 = 3/1-24,4000

EnableSyslog = 1
SyslogServerIP = 10.2.2.1

CallProgressTonesFilename = 'CPUSA.dat'
;CASFileName = 'E_M_WinkTable.dat'
SaveConfiguration = 1
```

6.6 Networking Parameters

Table 6-1: Networking Parameters (continues on pages 130 to 138)

ini File Field Name Web Parameter Name	Valid Range and Description
EthernetPhyConfiguration	0 = 10 Base-T half-duplex 1 = 10 Base-T full-duplex 2 = 100 Base-TX half-duplex 3 = 100 Base-TX full-duplex 4 = auto-negotiate (default) For detailed information on Ethernet interface configuration, refer to Section 9.1 on page 229.
DHCPEnable [Enable DHCP]	0 = Disable DHCP support on the gateway (default). 1 = Enable DHCP support on the gateway. After the gateway is powered up, it attempts to communicate with a BootP server. If a BootP server is not responding and if DHCP is enabled, then the gateway attempts to get its IP address and other network parameters from the DHCP server. **Web Note:** After you enable the DHCP Server (from the Web browser) follow this procedure: • Click the Submit button. • Save the configuration using the **Maintenance** button (before you reset the gateway). For information on how to save the configuration, refer to Section 5.9.2 on page 124. • Reset the gateway *directly* (Web reset doesn't trigger the BootP/DHCP procedure and the parameter DHCPEnable reverts to '0'). Note that throughout the DHCP procedure the BootP/TFTP application must be deactivated. Otherwise, the gateway receives a response from the BootP server instead of the DHCP server. **Note:** The DHCPEnable is a special 'Hidden' parameter. Once defined and saved in flash memory, its assigned value doesn't revert to its default even if the parameter doesn't appear in the *ini* file.
EnableLanWatchDog [Enable LAN Watchdog]	0 = Disable LAN Watch-Dog (default). 1 = Enable LAN Watch-Dog. When LAN Watch-Dog is enabled, the gateway's overall communication integrity is checked periodically. If no communication for about 3 minutes is detected, the gateway performs a self test. • If the self test succeeds, the problem is logical link down (i.e. Ethernet cable disconnected on the switch side), and the Busy out mechanism is activated if enabled (EnableBusyOut = 1). • If the self test fails, the gateway restarts to overcome internal fatal communication error. **Note:** Enable LAN Watchdog is relevant only if the Ethernet connection is full duplex.
DNSPriServerIP [DNS Primary Server IP]	IP address of the primary DNS server in dotted format notation.
DNSSecServerIP [DNS Secondary Server IP]	IP address of the secondary DNS server in dotted format notation.

Table 6-1: Networking Parameters (continues on pages 130 to 138)

ini File Field Name Web Parameter Name	Valid Range and Description
DNS2IP [Internal DNS Table]	Internal DNS table, used to resolve host names to IP addresses. Two different IP addresses (in dotted format notation) can be assigned to a hostname. DNS2IP = <Hostname>, <first IP address>, <second IP address> **Note 1:** If the internal DNS table is configured, the gateway first tries to resolve a domain name using this table. If the domain name isn't found, the gateway performs a DNS resolution using an external DNS server. **Note 2:** This parameter can appear up to 10 times.
EnableSTUN [Enable STUN]	0 = STUN protocol is disabled (default). 1 = STUN protocol is enabled. When enabled, the gateway functions as a STUN client and communicates with a STUN server located in the public internet. STUN is used to discover whether the gateway is located behind a NAT and the type of that NAT. In addition, it is used to determine the IP addresses and port numbers that the NAT assigns to outgoing signaling messages (using SIP) and media streams (using RTP, RTCP and T.38). STUN works with many existing NAT types, and does not require any special behavior from them. This parameter cannot be changed on-the-fly and requires a gateway reset.
STUNServerPrimaryIP [STUN Server Primary IP]	The IP address of the primary STUN server.
STUNServerSecondaryIP [STUN Server Secondary IP]	The IP address of the secondary STUN server.
STUNServerDomainName	Defines the domain name for the Simple Traversal of User Datagram Protocol (STUN) server's address (used for retrieving all STUN servers with an SRV query). The STUN client can perform the required SRV query to resolve this domain name to an IP address and port, sort the server list, and use the servers according to the sorted list. **Note:** Use either the STUNServerPrimaryIP or the STUNServerDomainName parameter, with priority to the first one.
NATBindingDefaultTimeout	Defines the default NAT binding lifetime in seconds. STUN is used to refresh the binding information after this time expires. The valid range is 0 to 2592000. The default value is 30.
DisableNAT	Enables / disables the Network Address Translation (NAT) mechanism. 0 = Enabled. 1 = Disabled (default). **Note:** The compare operation that is performed on the IP address is enabled by default and is controlled by the parameter 'EnableIPAddrTranslation'. The compare operation that is performed on the UDP port is disabled by default and is controlled by the parameter 'EnableUDPPortTranslation'.
EnableRport	Enables / disables the usage of the 'rport' parameter in the Via header. 0 = Enabled. 1 = Disabled (default). The gateway adds an 'rport' parameter to the Via header field of each outgoing SIP message. The first Proxy that receives this message sets the 'rport' value of the response to the actual port from which the request was received. This method is used, for example, to enable the gateway to identify its port mapping outside a NAT. If the Via doesn't include 'rport' tag, the destination port of the response will be taken from the host part of the VIA. If the Via includes 'rport' tag with no port value, the destination port of the response will be the source port of the incoming request. If the Via includes 'rport' tag with a port value (rport=1001), the destination port of the response will be the port indicated in the 'rport' tag.

Table 6-1: Networking Parameters (continues on pages 130 to 138)

ini File Field Name Web Parameter Name	Valid Range and Description
EnableIPAddrTranslation	0 = Disable IP address translation. 1 = Enable IP address translation for RTP, RTCP and T.38 packets (default). 2 = Enable IP address translation for ThroughPacket™. 3 = Enable IP address translation for all protocols (RTP, RTCP, T38 and ThroughPacket™). When enabled, the gateway compares the source IP address of the first incoming packet, to the remote IP address stated in the opening of the channel. If the two IP addresses don't match, the NAT mechanism is activated. Consequently, the remote IP address of the outgoing stream is replaced by the source IP address of the first incoming packet. **Note:** The NAT mechanism must be enabled for this parameter to take effect (DisableNAT = 0).
EnableUDPPortTranslation	0 = Disable UDP port translation (default). 1 = Enable UDP port translation. When enabled, the gateway compares the source UDP port of the first incoming packet, to the remote UDP port stated in the opening of the channel. If the two UDP ports don't match, the NAT mechanism is activated. Consequently, the remote UDP port of the outgoing stream is replaced by the source UDP port of the first incoming packet. **Note:** The NAT mechanism and the IP address translation must be enabled for this parameter to take effect (DisableNAT = 0, EnableIpAddrTranslation = 1).
NoOperationSendingMode	Enables or disables the transmission of RTP or T.38 No-Op packets. Valid options include: 0 = Disable (default) 1 = Enable This mechanism ensures that the NAT binding remains open during RTP or T.38 silence periods.
RTPNoOpEnable	Obsolete parameter; use NoOperationSendingMode instead.
NoOpInterval	Defines the time interval in which RTP or T.38 No-Op packets are sent in the case of silence (no RTP / T.38 traffic) when No-Op packet transmission is enabled. The valid range is 20 to 65,000 msec. The default is 10,000. **Note:** To enable No-Op packet transmission, use the NoOperationSendingMode parameter.
RTPNoOpInterval	Obsolete parameter; use NoOpInterval instead.
RTPNoOpPayloadType	Determines the payload type of No-Op packets. the valid range is 96 to 127. The default value is 120.
EnableDetectRemoteMACChange	Changes the RTP packets according to the MAC address of received RTP packets and according to Gratuitous Address Resolution Protocol (GARP) messages. Valid options include: 0 = nothing is changed. 1 = If the gateway receives RTP packets with a different source MAC address (than the MAC address of the transmitted RTP packets), then it sends RTP packets to this MAC address and removes this IP entry from the gateway's ARP cache table. 2 = The gateway uses the received GARP packets to change the MAC address of the transmitted RTP packets. 3 = both 1 and 2 options above are used (default).
StaticNatIP [NAT IP Address]	Static NAT IP address. Global gateway IP address. Define if static Network Address Translation (NAT) device is located between the gateway and the Internet.

Table 6-1: Networking Parameters (continues on pages 130 to 138)

ini File Field Name Web Parameter Name	Valid Range and Description
SyslogServerIP [Syslog Server IP Address]	IP address (in dotted format notation) of the computer you are using to run the Syslog Server. The Syslog Server is an application designed to collect the logs and error messages generated by the VoIP gateway. **Note:** Use the SyslogServerPort parameter to define the Syslog server's port. For information on the Syslog server, refer to Section 14.2 on page 304.
SyslogServerPort [Syslog Server Port]	Defines the UDP port of the Syslog server. The valid range is 0 to 65,535. The default port value is 514. For information on the Syslog server, refer to Section 14.2 on page 304.
EnableSyslog [Enable Syslog]	Sends the logs and error message generated by the gateway to the Syslog Server. 0 = Disable (logs and errors are not sent to the Syslog Server -- default). 1 = Enable. **Note1:** If you enable Syslog (i.e, EnableSyslog = 1), you must enter an IP address and a port number using SyslogServerIP and SyslogServerPort parameters. **Note 2:** Syslog messages may increase the network traffic. **Note 3:** To configure the Syslog logging levels use the parameter 'GwDebugLevel'.
BaseUDPport [RTP Base UDP Port]	Lower boundary of UDP port used for RTP, RTCP (Real-Time Control Protocol) (RTP port + 1) and T.38 (RTP port + 2). The upper boundary is the Base UDP Port + 10 * (number of gateway's channels). The range of possible UDP ports is 6,000 to 64,000. The default base UDP port is 6000. For example: If the Base UDP Port is set to 6000 (the default) then: The first channel uses the following ports: RTP 6000, RTCP 6001 and T.38 6002; the second channel uses: RTP 6010, RTCP 6011 and T.38 6012, etc. **Note:** If RTP Base UDP Port is not a factor of 10, the following message is generated: 'invalid local RTP port'. For detailed information on the default RTP/RTCP/T.38 port allocation, refer to Section E.3 on page 366.
RemoteBaseUDPPort [Remote RTP Base UDP Port]	Determines the lower boundary of UDP ports used for RTP, RTCP and T.38 by a remote gateway. If this parameter is set to a non-zero value, ThroughPacket™ is enabled. Note that the value of 'RemoteBaseUDPPort' on the local gateway must equal the value of 'BaseUDPPort' of the remote gateway. The gateway uses these parameters to identify and distribute the payloads from the received multiplexed IP packet to the relevant channels. The valid range is the range of possible UDP ports: 6,000 to 64,000. The default value is 0 (ThroughPacket™ is disabled). **Note:** To enable ThroughPacket™ the parameters 'L1L1ComplexTxUDPPort' and 'L1L1ComplexRxUDPPort' must be set to a non-zero value. For detailed information on ThroughPacket™, refer to Section 8.4 on page 212.
L1L1ComplexTxUDPPort [RTP Multiplexing Local UDP Port]	Determines the local UDP port used for outgoing multiplexed RTP packets (applies to the ThroughPacket™ mechanism). The valid range is the range of possible UDP ports: 6,000 to 64,000. The default value is 0 (ThroughPacket™ is disabled). This parameter cannot be changed on-the-fly and requires a gateway reset.

Table 6-1: Networking Parameters (continues on pages 130 to 138)

ini File Field Name Web Parameter Name	Valid Range and Description
L1L1ComplexRxUDPPort [RTP Multiplexing Remote UDP Port]	Determines the remote UDP port the multiplexed RTP packets are sent to, and the local UDP port used for incoming multiplexed RTP packets (applies to the ThroughPacket™ mechanism). The valid range is the range of possible UDP ports: 6,000 to 64,000. The default value is 0 (ThroughPacket™ is disabled). This parameter cannot be changed on-the-fly and requires a gateway reset. **Note:** All gateways that participate in the same ThroughPacket™ session must use the same 'L1L1ComplexRxUDPPort'.
NTPServerIP [NTP Server IP Address]	IP address (in dotted format notation) of the NTP server. The default IP address is 0.0.0.0 (the internal NTP client is disabled). For information on NTP support, refer to Section 9.8 on page 236.
NTPServerUTCOffset [NTP UTC Offset]	Defines the UTC (Universal Time Coordinate) offset (in seconds) from the NTP server. The default offset is 0. The offset range is –43200 to 43200 seconds.
NTPUpdateInterval [NTP Update Interval]	Defines the time interval (in seconds) the NTP client requests for a time update. The default interval is 86400 seconds (24 hours). The range is 0 to 214783647 seconds. **Note:** It isn't recommended to be set beyond one month (2592000 seconds).
IP Routing Table parameters: The IP routing *ini* file parameters are array parameters. Each parameter configures a specific column in the IP routing table. The first entry in each parameter refers to the first row in the IP routing table, the second entry to the second row and so forth. In the following example two rows are configured when the gateway is in network 10.31.x.x: RoutingTableDestinationsColumn = 130.33.4.6, 83.4.87.6 RoutingTableDestinationMasksColumn = 255.255.255.255, 255.255.255.0 RoutingTableGatewaysColumn = 10.31.0.1, 10.31.0.112 RoutingTableInterfacesColumn = 0, 1 RoutingTableHopsCountColumn = 20, 20	
RoutingTableDestinationsColumn	Specifies the IP address of the destination host / network.
RoutingTableDestinationMasksColumn	Specifies the subnet mask of the destination host / network.
RoutingTableGatewaysColumn	Specifies the IP address of the router to which the packets are sent if their destination matches the rules in the adjacent columns.
RoutingTableHopsCountColumn	The maximum number of allowed routers between the gateway and destination.
RoutingTableInterfacesColumn	Specifies the network type the routing rule is applied to. 0 = OAM (default). 1 = Control. 2 = Media.
VLAN Parameters	
VlanMode [VLAN Mode]	Sets the VLAN functionality. 0 = Disable (default) 1 = Enable 2 [PassThrough] = N/A.
VlanNativeVlanID [Native VLAN ID]	Sets the native VLAN identifier (PVID, Port VLAN ID). The valid range is 1 to 4094. The default value is 1.
VlanOamVlanID [OAM VLAN ID]	Sets the OAM (Operation, Administration and Management) VLAN identifier. The valid range is 1 to 4094. The default value is 1.
VlanControlVlanID [Control VLAN ID]	Sets the control VLAN identifier. The valid range is 1 to 4094. The default value is 2.
VlanMediaVlanID [Media VLAN ID]	Sets the media VLAN identifier. The valid range is 1 to 4094. The default value is 3.

Table 6-1: Networking Parameters (continues on pages 130 to 138)

ini File Field Name Web Parameter Name	Valid Range and Description
VlanNetworkServiceClassPriority [Network Priority]	Sets the priority for Network service class content. The valid range is 0 to 7. The default value is 7.
VlanPremiumServiceClassMediaPriority [Media Premium Priority]	Sets the priority for the Premium service class content and media traffic. The valid range is 0 to 7. The default value is 6.
VlanPremiumServiceClassControlPriority [Control Premium Priority]	Sets the priority for the Premium service class content and control traffic. The valid range is 0 to 7. The default value is 6.
VlanGoldServiceClassPriority [Gold Priority]	Sets the priority for the Gold service class content. The valid range is 0 to 7. The default value is 4.
VlanBronzeServiceClassPriority [Bronze Priority]	Sets the priority for the Bronze service class content. The valid range is 0 to 7. The default value is 2.
EnableDNSasOAM	This parameter applies to both Multiple IPs and VLAN mechanisms. Multiple IPs: Determines the network type for DNS services. VLAN: Determines the traffic type for DNS services. 1 = OAM (default) 0 = Control.
EnableNTPasOAM	This parameter applies to both Multiple IPs and VLAN mechanisms. Multiple IPs: Determines the network type for NTP services. VLAN: Determines the traffic type for NTP services. 1 = OAM (default) 0 = Control.
VlanSendNonTaggedOnNative	Specify whether to send non-tagged packets on the native VLAN. 0 = Sends priority tag packets (default). 1 = Sends regular packets (with no VLAN tag).
Multiple IPs Parameters	
EnableMultipleIPs [IP Networking Mode]	Enables / disables the Multiple IPs mechanism. 0 = Disabled (default). 1 = Enabled.
LocalMediaIPAddress [IP Address]	The gateway's source IP address in the Media network. The default value is 0.0.0.0.
LocalMediaSubnetMask [Subnet Mask]	The gateway's subnet mask in the Media network. The default subnet mask is 0.0.0.0.
LocalMediaDefaultGW [Default Gateway Address]	The gateway's default gateway IP address in the Media network. The default value is 0.0.0.0.
LocalControlIPAddress [IP Address]	The gateway's source IP address in the Control network. The default value is 0.0.0.0.
LocalControlSubnetMask [Subnet Mask]	The gateway's subnet mask in the Control network. The default subnet mask is 0.0.0.0.
LocalControlDefaultGW [Default Gateway Address]	N/A. Use the IP Routing table instead (Advanced Configuration > Network Settings).
LocalOAMIPAddress [IP Address]	The gateway's source IP address in the OAM network. The default value is 0.0.0.0.
LocalOAMSubnetMask [Subnet Mask]	The gateway's subnet mask in the OAM network. The default subnet mask is 0.0.0.0.
LocalOAMDefaultGW [Default Gateway Address]	N/A. Use the IP Routing table instead (Advanced Configuration > Network Settings).
PPPoE Parameters	
EnablePPPoE	Enables the PPPoE (Point-to-Point Protocol over Ethernet) feature. 0 = Disable (default) 1 = Enable

Table 6-1: Networking Parameters (continues on pages 130 to 138)

ini File Field Name Web Parameter Name	Valid Range and Description
PPPoEUserName	User Name for PAP or Host Name for CHAP authentication. The valid range is a string of up to 47 characters. The default value is 0.
PPPoEPassword	Password for PAP or Secret for CHAP authentication. The valid range is a string of up to 47 characters. The default value is 0.
PPPoEServerName	Server Name for CHAP authentication. The valid range is a string of up to 47 characters. The default value is 0.
PPPoEStaticIPAddress	IP address to use in a static configuration setup. If set, used during PPP negotiation to request this specific IP address from the PPP server. If approved by the server, this IP address is used during the session. The valid IP address range is in dotted notation xxx.xxx.xxx.xxx. The default value is 0.0.0.0.
PPPoERecoverIPAddress	IP address to use when booting from the flash to non-PPPoE (Point-to-Point Protocol over Ethernet) environments. The valid IP address range is in dotted notation xxx.xxx.xxx.xxx. The default value is 10.4.10.4.
PPPoERecoverSubnetMask	Subnet Mask to use when booting from the flash to non-PPPoE (Point-to-Point Protocol over Ethernet) environments. The valid IP address range is in dotted notation xxx.xxx.xxx.xxx. The default value is 255.255.0.0.
PPPoERecoverDfgwAddress	Default GW address to use when booting from the flash to non-PPPoE (Point-to-Point Protocol over Ethernet) environments. The valid IP address range is in dotted notation xxx.xxx.xxx.xxx. The default value is 10.4.10.1.
PPPoELCPEchoEnable	Enables or disables the Point-to-Point Protocol over Ethernet (PPPoE) disconnection auto-detection feature. Valid options include: 0 = Disable 1 = Enable (default) By default, the PPPoE Client (i.e., embedded in the gateway) sends LCP Echo packets to the server to check that the PPPoE connection is open. Some Access Concentrators (PPPoE servers) don't reply to these LCP Echo requests, resulting in a disconnection. By disabling the LCP disconnection auto-detection feature, the PPPoE Client doesn't send LCP Echo packets to the server (and does not detect PPPoE disconnections).
Differential Services. For detailed information on IP QoS via Differentiated Services, refer to Section 9.9 on page 236.	
NetworkServiceClassDiffServ [Network QoS]	Sets the DiffServ value for Network service class content. The valid range is 0 to 56. The default value is 48.
PremiumServiceClassMediaDiffServ [Media Premium QoS]	Sets the DiffServ value for Premium Media service class content (only if IPDiffServ is not set in the selected IP Profile). The valid range is 0 to 56. The default value is 46. **Note:** The value for the Premium Control DiffServ is determined by (according to priority): (1) IPDiffServ value in the selected IP Profile. (2) PremiumServiceClassMediaDiffServ.
PremiumServiceClassControlDiffServ [Control Premium QoS]	Sets the DiffServ value for Premium Control service class content (only if ControlIPDiffserv is not set in the selected IP Profile). The valid range is 0 to 56. The default value is 46. **Note:** The value for the Premium Control DiffServ is determined by (according to priority): (1) ControlPDiffserv value in the selected IP Profile. (2) PremiumServiceClassControlDiffServ.
GoldServiceClassDiffServ [Gold QoS]	Sets the DiffServ value for the Gold service class content. The valid range is 0 to 56. The default value is 26.

Table 6-1: Networking Parameters (continues on pages 130 to 138)

ini File Field Name Web Parameter Name	Valid Range and Description
BronzeServiceClassDiffServ [Bronze QoS]	Sets the DiffServ value for the Bronze service class content. The valid range is 0 to 56. The default value is 10.
NFS Table Parameters (NFSServers). For an NFS *ini* file example, refer to Figure 5-20 on page 86.	
NFSServers_Index [Line Number]	The row index of the remote file system. The valid range is 0 to 4.
NFSServers_HostOrIP [Host / IP]	The domain name or IP address of the NFS server. If a domain name is provided, a DNS server must be configured.
NFSServers_RootPath [Root Path]	Path to the root of the remote file system. In the format: '/' + [path] For example, /audio
The combination of Host / IP and Root Path must be unique for each row in the table. For example, there must be only one row in the table with a Host / IP of 192.168.1.1 and Root Path of /audio.	
NFSServers_NfsVersion [NFS Version]	NFS version to use with the remote file system, 2 or 3 (default).
NFSServers_AuthType [Auth Type]	Identifies the authentication method used with the remote file system. AUTH_NULL **[0]**. AUTH_UNIX **[1]** (default).
NFSServers_UID [UID]	User ID used in authentication if using AUTH_UNIX. The valid range is 0 to 65537. The default is 0.
NFSServers_GID [GID]	Group ID used in authentication if using AUTH_UNIX. The valid range is 0 to 65537. The default is 1
NFSServers_VlanType [VLAN Type]	The VLAN, OAM **[0]** or Media **[1]**, to use when accessing the remote file system. The default is to use the media VLAN. This parameter applies only if VLANs are enabled or if Multiple IPs is configured (refer to Section 9.10 on page 237).
Internal Firewall	
AccessList_Source_IP [Source IP]	IP address (or DNS name) of source network, or a specific host.
AccessList_Net_Mask [Mask]	IP network mask. 255.255.255.255 for a single host or the appropriate value for the source IP addresses. The IP address of the sender of the incoming packet is bitwise ANDed with this mask and then compared to the field 'Source IP'.
AccessList_Start_Port **AccessList_End_Port** [Local Port Range]	The destination UDP/TCP ports (on this device) to which packets are sent. The valid range is 0 to 65535. **Note:** When the protocol type isn't TCP or UDP, the entire range must be provided.
AccessList_Protocol [Protocol]	The protocol type (e.g., UDP, TCP, ICMP, ESP or 'Any'), or the IANA protocol number (in the range of 0 (Any) to 255). **Note:** The protocol field also accepts the abbreviated strings 'SIP', 'MGCP', 'MEGACO' and 'HTTP'. Specifying these strings implies selection of the TCP or UDP protocols, and the appropriate port numbers as defined on the device.
AccessList_Packet_Size [Packet Size]	Maximum allowed packet size. The valid range is 0 to 65535. **Note:** When filtering fragmented IP packets, the Packet Size field relates to the overall (reassembled) packet size, not to the size of each fragment.
AccessList_Byte_Rate [Byte Rate]	Expected traffic rate (bytes per second).
AccessList_Byte_Burst Burst Bytes	Tolerance of traffic rate limit (number of bytes)
AccessList_Allow_Type [Action Upon Match]	Action upon match (allow or block)
AccessList_MatchCount [Match Count]	A read-only field that provides the number of packets accepted / rejected by a specific rule.

6.7 System Parameters

Table 6-2: System Parameters (continues on pages 138 to 143)

ini File Field Name Web Parameter Name	Valid Range and Description
EnableDiagnostics	Checks the correct functionality of the different hardware components on the gateway. On completion of the check, if the test fails, the gateway sends information on the test results of each hardware component to the Syslog server. 0 = Rapid and Enhanced self-test mode (default). 1 = Detailed self-test mode (full test of DSPs, PCM, Switch, LAN, PHY and Flash). 2 = A quicker version of the Detailed self-test mode (full test of DSPs, PCM, Switch, LAN, PHY, but partial test of Flash). For detailed information, refer to Section 14.1 on page 303.
GWAppDelayTime [Delay After Reset [sec]]	Defines the amount of time (in seconds) the gateway's operation is delayed after a reset cycle. The valid range is 0 to 45. The default value is 7 seconds. **Note:** This feature helps to overcome connection problems caused by some LAN routers or IP configuration parameters change by a DHCP Server.
ActivityListToLog [Activity Types to Report via 'Activity Log' Messages]	The Activity Log mechanism enables the gateway to send log messages (to a Syslog server) that report certain types of web actions according to a pre-defined filter. The following filters are available: **PVC** (Parameters Value Change) - Changes made on-the-fly to parameters. **AFL** (Auxiliary Files Loading) - Loading of auxiliary files (e.g., via Certificate screen). **DR** (Device Reset) - Device reset via the Maintenance screen. **FB** (Flash Memory Burning) - Burning of files / parameters to flash (e.g., Maintenance screen). **SWU** (Device Software Update) - cmp loading via the Software Upgrade Wizard. **ARD** (Access to Restricted Domains) - Access to Restricted Domains. The following screens are restricted: (1) ini parameters (AdminPage) (2) General Security Settings (3) Configuration File (4) IPSec/IKE tables (5) Software Upgrade Key (6) Internal Firewall (7) Web Access List. (8) Web User Accounts **NAA** (Non Authorized Access) - Attempt to access the Embedded Web Server with a false / empty username or password. **SPC** (Sensitive Parameters Value Change) - Changes made to sensitive parameters: (1) IP Address (2) Subnet Mask (3) Default Gateway IP Address (4) ActivityListToLog For example: ActivityListToLog = 'pvc', 'afl', 'dr', 'fb', 'swu', 'ard', 'naa', 'spc'

Table 6-2: System Parameters (continues on pages 138 to 143)

ini File Field Name Web Parameter Name	Valid Range and Description
MaxEchoCancellerLength [Max Echo Canceller Length] **Note:** It isn't necessary to configure the parameter EchoCancellerLength as it automatically acquires its value from the parameter MaxEchoCancellerLength.	Maximum Echo Canceler Length in msec: 0 = based on various internal gateway settings -- 64 msec (default) 4 = 32 msec 11 = 64 msec 22 = 128 msec **Note 1:** When set to 128 msec, the number of available gateway channels is reduced by a factor of 5/6. For example: Gateway with 8 E1 spans capacity is reduced to 6 spans (180 channels), while gateway with 8 T1 spans capacity remains the same (192 channels). **Note 2:** The gateway must be reset after the value of 'MaxEchoCancellerLength' is changed.
ECHybridLoss	Sets the four wire to two wire worst case Hybrid loss, the ratio between the signal level sent to the hybrid and the echo level returning from the hybrid. 0 = 6 dB (default) 1 = 9 dB 2 = 0 dB 3 = 3 dB
GwDebugLevel [Debug Level]	Defines the Syslog logging level (usually set to 5 if debug traces are needed). 0 = Debug is disabled (default) 1 = Flow debugging is enabled 2 = Flow and board interface debugging are enabled 3 = Flow, board interface and stack interface debugging are enabled 4 = Flow, board interface, stack interface and session manager debugging are enabled 5 = Flow, board interface, stack interface, session manager and board interface expanded debugging are enabled
CDRReportLevel [CDR Report Level]	Determines whether or not CDRs are sent to the Syslog server, and if enabled, at which events they are sent. Valid options include: 0 = Call Detail Record (CDR) is not used. 1 = CDR is sent to the Syslog server at the end of each call. 2 = CDR report is sent to Syslog at the start and end of each call. 3 = CDR report is sent to Syslog at connection and at the end of each call. The CDR Syslog message complies with RFC 3161 and is identified by: Facility = 17 (local1) and Severity = 6 (Informational) **Note:** This parameter replaces the EnableCDR parameter.
CDRSyslogServerIP [CDR Server IP Address]	Defines the destination IP address for CDR logs. The default value is a null string that causes the CDR messages to be sent with all Syslog messages. **Note:** The CDR messages are sent to UDP port 514 (default Syslog port).
HeartBeatDestIP	Destination IP address (in dotted format notation) to which the gateway sends proprietary UDP 'ping' packets. The default IP address is 0.0.0.0.
HeartBeatDestPort	Destination UDP port to which the heartbeat packets are sent. The range is 0 to 64000. The default is 0.
HeartBeatIntervalmsec	Delay (in msec) between consecutive heartbeat packets. 10 = 100000. -1 = disabled (default).

Table 6-2: System Parameters (continues on pages 138 to 143)

ini File Field Name Web Parameter Name	Valid Range and Description
EnableRAI	0 = Disable RAI (Resource Available Indication) service (default). 1 = Enable RAI service. If RAI is enabled, an SNMP 'acBoardCallResourcesAlarm' Alarm Trap is sent if gateway resources fall below a predefined (configurable) threshold.
RAIHighThreshold	High Threshold (in percentage) that defines the gateway's busy endpoints. The range is 0 to 100. The default value is 90%. When the percentage of the gateway's busy endpoints exceeds the value configured in High Threshold, the gateway sends an SNMP 'acBoardCallResourcesAlarm' Alarm Trap with a 'major' Alarm Status. **Note:** The gateway's available Resources are calculated by dividing the number of busy endpoints by the total number of available gateway endpoints.
RAILowThreshold	Low Threshold (in percentage) that defines the gateway's busy endpoints. The range is 0 to 100. The default value is 90%. When the percentage of the gateway's busy endpoints falls below the value defined in Low Threshold, the gateway sends an SNMP 'acBoardCallResourcesAlarm' Alarm Trap with a 'cleared' Alarm Status.
RAILoopTime	Time interval (in seconds) that the gateway checks for resource availability. The default is 10 seconds.
Disconnect Supervision Parameters	
DisconnectOnBrokenConnection [Disconnect on Broken Connection]	0 = Don't release the call. 1 = Call is released If RTP packets are not received for a predefined timeout (default). **Note 1:** If enabled, the timeout is set by the parameter 'BrokenConnectionEventTimeout', in 100 msec resolution. The default timeout is 10 seconds: (BrokenConnectionEventTimeout =100). **Note 2:** This feature is applicable only if RTP session is used without Silence Compression. If Silence Compression is enabled, the gateway doesn't detect that the RTP connection is broken. **Note 3:** During a call, if the source IP address (from where the RTP packets were sent) is changed without notifying the gateway, the gateway filters these RTP packets. To overcome this issue, set 'DisconnectOnBrokenConnection=0'; the gateway doesn't detect RTP packets arriving from the original source IP address, and switches (after 300 msec) to the RTP packets arriving from the new source IP address.
BrokenConnectionEventTimeout [Broken Connection Timeout]	The amount of time (in 100 msec units) an RTP packets isn't received, after which a call is disconnected. The valid range is 1 to 1000. The default value is 100 (10 seconds). **Note 1:** Applicable only if 'DisconnectOnBrokenConnection = 1'. **Note 2:** Currently this feature works only if Silence Suppression is disabled.
EnableSilenceDisconnect [Disconnect Call on Silence Detection]	1 = The gateway disconnect calls in which silence occurs in both (call) directions for more than 120 seconds. 0 = Call is not disconnected when silence is detected (default). The silence duration can be set by the 'FarEndDisconnectSilencePeriod' parameter (default 120). **Note:** To activate this feature set: 'EnableSilenceCompression' to 1 and 'FarEndDisconnectSilenceMethod' to 1.
FarEndDisconnectSilencePeriod [Silence Detection Period]	Duration of silence period (in seconds) prior to call disconnection. The range is 10 to 28800 (8 hours). The default is 120 seconds. Applicable to gateways, that use DSP templates 2 or 3.

Table 6-2: System Parameters (continues on pages 138 to 143)

ini File Field Name Web Parameter Name	Valid Range and Description
FarEndDisconnectSilenceMethod [Silence Detection Method]	Silence detection method. 0 (None) = Silence detection option is disabled. 1 (Packets Count) = According to packet count. 2 (Voice/Energy Detectors) = N/A. 3 (All) = N/A.
FarEndDisconnectSilenceThreshold	Threshold of the packet count (in percents), below which is considered silence by the media gateway. The valid range is 1 to 100. The default is 8%. **Note:** Applicable only if silence is detected according to packet count (FarEndDisconnectSilenceMethod = 1).
Automatic Update Parameters	
CmpFileURL	Specifies the name of the *cmp* file and the location of the server (IP address or FQDN) from which the gateway loads a new *cmp* file and updates itself. The *cmp* file can be loaded using: HTTP, HTTPS, FTP, FTPS or NFS. For example: http://192.168.0.1/filename **Note 1:** When this parameter is set in the *ini* file, the gateway always loads the *cmp* file after it is reset. **Note 2:** The *cmp* file is validated before it is burned to flash. The checksum of the *cmp* file is also compared to the previously-burnt checksum to avoid unnecessary resets.
IniFileURL	Specifies the name of the *ini* file and the location of the server (IP address or FQDN) from which the gateway loads the *ini* file. The *ini* file can be loaded using: HTTP, HTTPS, FTP, FTPS or NFS. For example: http://192.168.0.1/filename http://192.8.77.13/config<MAC> https://<username>:<password>@<IP address>/<file name> **Note 1:** When using HTTP or HTTPS, the date and time of the *ini* file are validated. Only more recently-dated *ini* files are loaded. **Note 2:** The optional string '<MAC>' is replaced with the gateway's MAC address. Therefore, the gateway requests an *ini* file name that contains its MAC address. This option enables loading different configurations for specific gateways.
PrtFileURL	Specifies the name of the Prerecorded Tones file and the location of the server (IP address or FQDN) from which it is loaded. http://server_name/file, https://server_name/file.
CptFileURL	Specifies the name of the CPT file and the location of the server (IP address or FQDN) from which it is loaded. http://server_name/file, https://server_name/file.
CasFileURL	Specifies the name of the CAS file and the location of the server (IP address or FQDN) from which it is loaded. http://server_name/file, https://server_name/file.
VpFileURL	Specifies the name of the Voice Prompts file and the location of the server (IP address or FQDN) from which it is loaded. http://server_name/file, https://server_name/file.
UserInfoFileURL	Specifies the name of the User Information file and the location of the server (IP address or FQDN) from which it is loaded. http://server_name/file, https://server_name/file.
AutoUpdateCmpFile	Enables / disables the Automatic Update mechanism for the cmp file. 0 = The Automatic Update mechanism doesn't apply to the cmp file (default). 1 = The Automatic Update mechanism includes the cmp file.
AutoUpdateFrequency	Determines the number of minutes the gateway waits between automatic updates. The default value is 0 (the update at fixed intervals mechanism is disabled).

Table 6-2: System Parameters (continues on pages 138 to 143)

<table>
<tr><th>ini File Field Name
Web Parameter Name</th><th colspan="2">Valid Range and Description</th></tr>
<tr><td>AutoUpdatePredefinedTime</td><td colspan="2">Schedules an automatic update to a predefined time of the day.
The range is 'HH:MM' (24-hour format).
For example: 20:18
Note: The actual update time is randomized by five minutes to reduce the load on the Web servers.</td></tr>
<tr><td>ResetNow</td><td colspan="2">Invokes an immediate restart of the gateway.
This option can be used to activate offline (not on-the-fly) parameters that are loaded via IniFileUrl.
0 = The immediate restart mechanism is disabled (default).
1 = The gateway immediately restarts after an ini file with this parameter set to 1 is loaded.</td></tr>
<tr><td colspan="3">BootP and TFTP Parameters</td></tr>
<tr><td colspan="3">The BootP parameters are special ‘Hidden’ parameters. Once defined and saved in the flash memory, they are used even if they don't appear in the ini file.</td></tr>
<tr><td rowspan="2">BootPRetries</td><td colspan="2">This parameter is used to:
Note: This parameter only takes effect from the next reset of the gateway.</td></tr>
<tr><td>Set the number of BootP requests the gateway sends during start-up. The gateway stops sending BootP requests when either BootP reply is received or number of retries is reached.

1 = 1 BootP retry, 1 second.
2 = 2 BootP retries, 3 second.
3 = 3 BootP retries, 6 second (default).
4 = 10 BootP retries, 30 second.
5 = 20 BootP retries, 60 second.
6 = 40 BootP retries, 120 second.
7 = 100 BootP retries, 300 second.
15 = BootP retries indefinitely.</td><td>Set the number of DHCP packets the gateway sends.
After all packets were sent, if there's still no reply, the gateway loads from flash.

1 = 4 DHCP packets
2 = 5 DHCP packets
3 = 6 DHCP packets (default)
4 = 7 DHCP packets
5 = 8 DHCP packets
6 = 9 DHCP packets
7 = 10 DHCP packets
15 = 18 DHCP packets</td></tr>
<tr><td>BootPSelectiveEnable</td><td colspan="2">Enables the Selective BootP mechanism.
1 = Enabled.
0 = Disabled (default).

The Selective BootP mechanism (available from Boot version 1.92) enables the gateway’s integral BootP client to filter unsolicited BootP/DHCP replies (accepts only BootP replies that contain the text ‘AUDC’ in the vendor specific information field). This option is useful in environments where enterprise BootP/DHCP servers provide undesired responses to the gateway’s BootP requests.
Note: When working with DHCP (DHCPEnable = 1) the selective BootP feature must be disabled.</td></tr>
<tr><td>BootPDelay</td><td colspan="2">The interval between the device’s startup and the first BootP/DHCP request that is issued by the device.
1 = 1 second (default).
2 = 3 second.
3 = 6 second.
4 = 30 second.
5 = 60 second.
Note: This parameter only takes effect from the next reset of the device.</td></tr>
</table>

Table 6-2: System Parameters (continues on pages 138 to 143)

ini File Field Name Web Parameter Name	Valid Range and Description
ExtBootPReqEnable	0 = Disable (default). 1 = Enable extended information to be sent in BootP request. If enabled, the device uses the vendor specific information field in the BootP request to provide device-related initial startup information such as board type, current IP address, software version, etc. For a full list of the vendor specific Information fields, refer to Section 7.3 on page 205. The BootP/TFTP configuration utility displays this information in the 'Client Info' column (refer to Figure D-1). **Note:** This option is not available on DHCP servers.

6.8 Web and Telnet Parameters

Table 6-3: Web and Telnet Parameters (continues on pages 143 to 144)

ini File Field Name Web Parameter Name	Valid Range and Description
WebAccessList_x [Web and Telnet Access List Screen]	Defines up to ten IP addresses that are permitted to access the gateway's Web and Telnet interfaces. Access from an undefined IP address is denied. This security feature is inactive (the gateway can be accessed from any IP address) when the table is empty. For example: WebAccessList_0 = 10.13.2.66 WebAccessList_1 = 10.13.77.7 The default value is 0.0.0.0 (the gateway can be accessed from any IP address).
WebRADIUSLogin [Use RADIUS for Web/Telnet Login]	Uses RADIUS queries for Web and Telnet interface authentication. 0 = Disabled (default). 1 = Enabled. When enabled, logging to the gateway's Web and Telnet embedded servers is performed via a RADIUS server. The gateway contacts a predefined server and verifies the given username and password pair against a remote database, in a secure manner. **Note 1:** The parameter 'EnableRADIUS' must be set to 1. **Note 2:** RADIUS authentication requires HTTP basic authentication, meaning the username and password are transmitted in clear text over the network. Therefore, users are recommended to set the parameter 'HttpsOnly = 1' to force the use of HTTPS, since the transport is encrypted.
DisableWebTask	0 = Enable Web management (default). 1 = Disable Web management.
ResetWebPassword	Resets the username and password of the primary and secondary accounts to their defaults. 0 = Password and username retain their values (default). 1 = Password and username are reset (for the default username and password, refer to Table 4-1 on page 49). **Note:** The username and password cannot be reset from the Web (i.e., via AdminPage or by loading an *ini* file).
DisableWebConfig	0 = Enable changing parameters from Web (default). 1 = Operate Web Server in 'read only' mode.
HTTPport	HTTP port used for Web management (default = 80)

Table 6-3: Web and Telnet Parameters (continues on pages 143 to 144)

ini File Field Name Web Parameter Name	Valid Range and Description
Telnet Parameters	
TelnetServerEnable [Embedded Telnet Server]	Enables or disables the embedded Telnet server. Telnet is disabled by default for security reasons. 0 = Disable (default). 1 = Enable (Unsecured). 2 = Enable Secured (SSL).
TelnetServerPort [Telnet Server TCP Port]	Defines the port number for the embedded Telnet server. The valid range = valid port numbers. The default port is 23.
TelnetServerIdleDisconnect [Telnet Server Idle Timeout]	Sets the timeout for disconnection of an idle Telnet session (in minutes). When set to zero, idle sessions are not disconnected. The valid range is any value. The default value is 0.
Customizing the Web Appearance Parameters For detailed information on customizing the Web Interface, refer to Section 11.6 on page 257.	
UseProductName	0 = Disabled (default). 1 = Enabled. If enabled, the 'UserProductNane' text string is displayed instead of the default product name.
UserProductName	Text string that replaces the product name. The default is '<Mediant 2000, TP-1610, or TP-260>'. The string can be up to 29 characters.
UseWebLogo	0 = Logo image is used (default). 1 = Text string is used instead of a logo image. If enabled, AudioCodes' default logo (or any other logo defined by the 'LogoFileName' parameter) is replaced with a text string defined by the 'WebLogoText' parameter.
WebLogoText	Text string that replaces the logo image. The string can be up to 15 characters.
LogoWidth	Width (in pixels) of the logo image. **Note:** The optimal setting depends on the resolution settings. The default value is 441, which is the width of AudioCodes' displayed logo.
LogoFileName	Name of the image file containing the user's logo. File name can be up to 47 characters. The logo file name can be used to replace AudioCodes' default Web logo with a user defined logo. Use a gif, jpeg or jpg image file.
BkgImageFileName	Name of the file containing the user's background image. File name can be up to 47 characters. The background file can be used to replace AudioCodes' default background image with a user defined background. Use a gif, jpeg or jpg image file.

6.9 Security Parameters

Table 6-4: Security Parameter (continues on pages 145 to 146)

ini File Field Name Web Parameter Name	Valid Range and Description
EnableIPSec [Enable IP Security]	Enables / disables the Secure Internet Protocol (IPSec) on the gateway. 0 = Disable (default). 1 = Enable.
EnableMediaSecurity [Enable Media Security]	Enables or disables the Secure Real-Time Transport Protocol (SRTP). 0 = SRTP is disabled (default). 1 = SRTP is enabled. **Note 1:** SRTP is available only if DSPVersionTemplateNumber = 0 or 2. **Note 2:** Use of SRTP reduces the number of available channels. TP-1610, Mediant 2000, TP-260: Template 0, 200 channels are available (per TPM). Template 2, 120 channels are available (per TPM).
MediaSecurityBehaviour [Media Security Behavior]	Determines the gateway's mode of operation when SRTP is used (EnableMediaSecurity = 1). 0 (Prefer) = The gateway initiates encrypted calls. If negotiation of the cipher suite fails, an unencrypted call is established. Incoming calls that don't include encryption information are accepted. 1 (Must) = The gateway initiates encrypted calls. If negotiation of the cipher suite fails, the call is terminated. Incoming calls that don't include encryption information are rejected (default).
EnableSIPS [Enable SIPS]	Enables secured SIP (SIPS) connections over multiple hops. 0 = Disabled (default). 1 = Enabled. When SIPTransportType = 2 (TLS) and EnableSIPS is disabled, TLS is used for the next network hop only. When SIPTransportType = 2 (TLS) or 1 (TCP) and EnableSIPS is enabled, TLS is used through the entire connection (over multiple hops). **Note:** If SIPS is enabled and SIPTransportType = UDP, the connection fails.
TLSLocalSIPPort [SIP TLS Local Port]	Local TLS port used to receive SIP messages. The default value is 5061. **Note:** The value of 'TLSLocalSIPPort' must be different to the value of 'TCPLocalSIPPort'.
SIPSRequireClientCertificate	0 = The gateway doesn't require client certificate (default). 1 = The gateway (when acting as a server for the TLS connection) requires reception of client certificate to establish the TLS connection. **Note:** The SIPS certificate files can be changed using the parameters 'HTTPSCertFileName' and 'HTTPSRootFileName'.
Secure Hypertext Transport Protocol (HTTPS) Parameters	
HTTPSOnly [Secured Web Connection]	Determines the protocol types used to access the Embedded Web Server. 0 = HTTP and HTTPS (default). 1 = HTTPS only (unencrypted HTTP packets are blocked).
HTTPSPort	Determine the local Secured HTTPS port of the device. The valid range is 1 to 65535 (other restrictions may apply within this range). The default port is 443.
WebAuthMode [HTTP Authentication Mode]	Determines the authentication mode for the Embedded Web Server. 0 = Basic authentication (clear text) is used (default). 1 = Digest authentication (MD5) is used. 2 = Digest authentication (MD5) is used for HTTP, and basic authentication is used for HTTPS. Note that when RADIUS login is enabled (WebRADIUSLogin = 1), basic authentication is forced.

Table 6-4: Security Parameter (continues on pages 145 to 146)

ini File Field Name Web Parameter Name	Valid Range and Description
HTTPSRequireClientCertificate	Requires client certificates for HTTPS connection. The client certificate must be preloaded to the gateway, and its matching private key must be installed on the managing PC. Time and date must be correctly set on the gateway, for the client certificate to be verified. 0 = Client certificates are not required (default). 1 = Client certificates are required.
HTTPSRootFileName	Defines the name of the HTTPS trusted root certificate file to be loaded via TFTP. The file must be in base64-encoded PEM (Privacy Enhanced Mail) format. The valid range is a 47-character string. **Note:** This parameter is only relevant when the gateway is loaded via BootP/TFTP. For information on loading this file via the Embedded Web Server, refer to the Security section in the User's Manual.
HTTPSPkeyFileName [Security Settings > Certificates]	Defines the name of a private key file (in unencrypted PEM format) to be loaded from the TFTP server.
HTTPSCertFileName	Defines the name of the HTTPS server certificate file to be loaded via TFTP. The file must be in base64-encoded PEM format. The valid range is a 47-character string. **Note:** This parameter is only relevant when the gateway is loaded via BootP/TFTP. For information on loading this file via the Embedded Web Server, refer to the Security section in the User's Manual.
Internal Firewall Parameters	
AccessList_Source_IP [Source IP]	IP address (or DNS name) of source network, or a specific host.
AccessList_Net_Mask [Mask]	IP network mask. 255.255.255.255 for a single host or the appropriate value for the source IP addresses. The IP address of the sender of the incoming packet is bitwise ANDed with this mask and then compared to the field 'Source IP'.
AccessList_Start_Port **AccessList_End_Port** [Local Port Range]	The destination UDP/TCP ports (on this device) to which packets are sent. The valid range is 0 to 65535. **Note:** When the protocol type isn't TCP or UDP, the entire range must be provided.
AccessList_Protocol [Protocol]	The protocol type (e.g., UDP, TCP, ICMP, ESP or 'Any'), or the IANA protocol number (in the range of 0 (Any) to 255). **Note:** The protocol field also accepts the abbreviated strings 'SIP', 'MGCP', 'MEGACO' and 'HTTP'. Specifying these strings implies selection of the TCP or UDP protocols, and the appropriate port numbers as defined on the device.
AccessList_Packet_Size [Packet Size]	Maximum allowed packet size. The valid range is 0 to 65535. **Note:** When filtering fragmented IP packets, the Packet Size field relates to the overall (reassembled) packet size, not to the size of each fragment.
AccessList_Byte_Rate [Byte Rate]	Expected traffic rate (bytes per second).
AccessList_Byte_Burst [Burst Bytes]	Tolerance of traffic rate limit (number of bytes)
AccessList_Allow_Type [Action Upon Match]	Action upon match (allow or block)
AccessList_MatchCount [Match Count]	A read-only field that provides the number of packets accepted / rejected by a specific rule.

6.10 RADIUS Parameters

For detailed information on the supported RADIUS attributes, refer to Section 8.9 on page 218.

Table 6-5: RADIUS Parameters (continues on pages 147 to 148)

ini File Field Name Web Parameter Name	Valid Range and Description
EnableRADIUS [Enable RADIUS]	0 = RADIUS application is disabled (default). 1 = RADIUS application is enabled.
AAAIndications [AAA Indications]	0 = No indications (default). 3 = Accounting only
MaxRADIUSSessions [Max. RADIUS Sessions]	Number of concurrent calls that can communicate with the RADIUS server (optional). The valid range is 0 to 240. The default value is 240.
SharedSecret [Shared Secret]	'Secret' used to authenticate the gateway to the RADIUS server. It should be a cryptographically strong password.
RADIUSRetransmission [RADIUS Max. Retransmissions]	Number of retransmission retries. The valid range is 1 to 10. The default value is 3.
RadiusTO	The interval between each retry (measured in seconds). The valid range is 1 to 30. The default value is 10.
RADIUSAuthServerIP [RADIUS Authentication Server IP Address]	IP address of Authentication and Authorization server.
RADIUSAuthPort [RADIUS Authentication Port]	Port number of Authentication and Authorization server. The default value is 1645.
RADIUSAccServerIP [RADIUS Accounting Server IP Address]	IP address of accounting server.
RADIUSAccPort [RADIUS Accounting Port]	Port number of Radius accounting server. The default value is 1646.
RadiusAccountingType [RADIUS Accounting Type]	Determines when a RADIUS accounting report is issued. 0 = At the Release of the call only (default). 1 = At the Connect and Release of the call. 2 = At the Setup and Release of the call.
DefaultAccessLevel [Default Access Level]	Defines the default access level for the gateway when the RADIUS (authentication) response doesn't include an access level attribute. The valid range is 0 to 255. The default value is 200 (Security Administrator').
BehaviorUponRadiusTimeout [Device Behavior Upon RADIUS Timeout]	Defines the gateway's operation if a response isn't received from the RADIUS server after the 5 seconds timeout expires: 0 (Deny Access) = the gateway denies access to the Web and Telnet embedded servers. 1 (Verify Access Locally) = the gateway checks the local username and password (default).
RadiusLocalCacheMode [Local RADIUS Password Cache Mode]	Defines the gateway's mode of operation regarding the timer (configured by the parameter RadiusLocalCacheTimeout) that determines the validity of the username and password (verified by the RADIUS server). 0 (Absolute Expiry Timer) = when you access a Web screen, the timeout doesn't reset but rather continues decreasing. 1 (Reset Timer Upon Access) = upon each access to a Web screen, the timeout always resets (reverts to the initial value configured by RadiusLocalCacheTimeout).

Table 6-5: RADIUS Parameters (continues on pages 147 to 148)

ini File Field Name Web Parameter Name	Valid Range and Description
RadiusLocalCacheTimeout [Local RADIUS Password Cache Timeout]	Defines the time (in seconds) the locally stored username and password (verified by the RADIUS server) are valid. When this time expires, the username and password becomes invalid and a must re-verified with the RADIUS server. The valid range is 1 to 0xFFFFFF. -1 = Never expires. 0 = Each request requires RADIUS authentication. The default value is 300 (5 minutes).
RadiusVSAVendorID [RADIUS VSA Vendor ID]	Defines the vendor ID the gateway accepts when parsing a RADIUS response packet. The valid range is 0 to 0xFFFFFFFF. The default value is 5003.
RadiusVSAAccessAttribute [RADIUS VSA Access Level Attribute]	Defines the code that indicates the access level attribute in the Vendor Specific Attributes (VSA) section of the received RADIUS packet. The valid range is 0 to 255. The default value is 35.

6.11 SNMP Parameters

Table 6-6: SNMP Parameters (continues on pages 148 to 149)

ini File Field Name Web Parameter Name	Valid Range and Description
DisableSNMP [Enable SNMP]	0 = SNMP is enabled (default). 1 = SNMP is disabled and no traps are sent.
SNMPPort	The device's local UDP port used for SNMP Get/Set commands. The range is 100 to 3999. The default port is 161.
SNMPTrustedMGR_x	Up to five IP addresses of remote trusted SNMP managers from which the SNMP agent accepts and processes get and set requests. **Note 1:** If no values are assigned to these parameters any manager can access the device. **Note 2:** Trusted managers can work with *all* community strings.
AlarmHistoryTableMaxSize	Determines the maximum number of rows in the Alarm History table. The parameter can be controlled by the Config Global Entry Limit MIB (located in the Notification Log MIB). The valid range is 50 to 1000. The default value is 500.
SNMP Trap Parameters	
SNMPManagerTableIP_x [SNMP Managers Table]	Up to five IP addresses of remote hosts that are used as SNMP Managers. The device sends SNMP traps to these IP addresses. Enter the IP address in dotted format notation, for example 108.10.1.255.
SNMPManagerTrapPort_x [SNMP Managers Table]	Up to five parameters used to define the Port numbers of the remote SNMP Managers. The device sends SNMP traps to these ports. **Note:** The first entry (out of the five) replaces the obsolete parameter SNMPTrapPort. The default SNMP trap port is 162 The valid SNMP trap port range is 100 to 4000.
SNMPManagerTrapUser_x	This parameter can be set to the name of any configured SNMPV3 user to associate with this trap destination. This determines the trap format, authentication level, and encryption level. By default, the trap is associated with the SNMP trap community string.
SNMPManagerIsUsed_x [SNMP Managers Table]	Up to five parameters, each determines the **validity** of the parameters (IP address and port number) of the corresponding SNMP Manager used to receive SNMP traps. 0 = Disabled (default) 1 = Enabled

Table 6-6: SNMP Parameters (continues on pages 148 to 149)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
SNMPManagerTrapSendingEnable _x [SNMP Managers Table]	Up to five parameters, each determines the activation/deactivation of sending traps to the corresponding SNMP Manager. 0 = Sending is disabled 1 = Sending is enabled (default)
SNMPTrapManagerHostName [Trap Manager Host Name]	Defines a FQDN of a remote host that is used as an SNMP Manager. The resolved IP address replaces the last entry in the trap manager table (defined by the parameter 'SNMPManagerTableIP_x') and the last trap manager entry of snmpTargetAddrTable in the snmpTargetMIB. For example: 'mngr.corp.mycompany.com'. The valid range is a 99-character string
SNMP Community String Parameters	
SNMPReadOnlyCommunityString _x [SNMP Community Strings Table]	Up to five read-only community strings (up to 19 characters each). The default string is 'public'.
SNMPReadWriteCommunityString _x [SNMP Community Strings Table]	Up to five read / write community strings (up to 19 characters each). The default string is 'private'.
SNMPTrapCommunityString [SNMP Community Strings Table]	Community string used in traps (up to 19 characters). The default string is 'trapuser'.
SNMP v3 Users Parameters	
SNMPUsers_Index [Index]	This is the table index. Its valid range is 0 to 9.
[SNMPUsers_Username] [Username]	Name of the SNMP v3 user. This name must be unique.
[SNMPUsers_AuthProtocol] [AuthProtocol]	Authentication protocol to be used for the SNMP v3 user. 0 = none (default) 1 = MD5 2 = SHA-1
[SNMPUsers_PrivProtocol] [PrivProtocol]	Privacy protocol to be used for the SNMP v3 user. 0 = none (default) 1 = DES 2 = 3DES 3 = AES128 4 = AES192 5 = AES256
[SNMPUsers_AuthKey] [AuthKey]	Authentication key. Keys can be entered in the form of a text password or long hex string. Keys are always persisted as long hex strings and keys are localized.
[SNMPUsers_PrivKey] [PrivKey]	Privacy key. Keys can be entered in the form of a text password or long hex string. Keys are always persisted as long hex strings and keys are localized.
[SNMPUsers_Group] [Group]	The group with which the SNMP v3 user is associated. 0 = read-only group (default) 1 = read-write group 2 = trap group **Note:** all groups can be used to send traps.

6.12 SIP Configuration Parameters

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
SIPTransportType [SIP Transport Layer]	Determines the *default* transport layer used for outgoing SIP calls initiated by the gateway. 0 = UDP (default). 1 = TCP. 2 = TLS (SIPS). **Note:** It is recommended to use TLS to communicate with a SIP Proxy and not for direct gateway-gateway communication.
TCPLocalSIPPort [SIP TCP Local Port]	Local TCP port used to receive SIP messages. The default value is 5060.
SIPDestinationPort [SIP Destination Port]	SIP destination port for sending initial SIP requests. The valid range is 1 to 65534. The default port is 5060. **Note:** SIP responses are sent to the port specified in the Via header.
EnableTCPConnectionReuse [Enable TCP Connection Reuse]	Enables the reuse of the same TCP connection for all calls to the same destination. Valid options include: 0 = Use a separate TCP connection for each call (default) 1 = Use the same TCP connection for all calls
LocalSIPPort [SIP Local Port]	Local UDP port used to receive SIP messages. The valid range is 1 to 65534. The default port is 5060.
SIPGatewayName [Gateway Name]	Use this parameter to assign a name to the gateway (e.g., 'gateway1.com'). Ensure that the name you choose is the one that the Proxy is configured with to identify your media gateway. **Note:** If specified, the gateway Name is used as the host part of the SIP URI in the From header. If not specified, the gateway IP address is used instead (default).
IsProxyUsed [Enable Proxy]	0 = Proxy isn't used, the internal routing table is used instead (default). 1 = Proxy is used.
ProxyIP [Proxy IP Address]	IP address (and optionally port number) of the primary Proxy server you are using. Enter the IP address as FQDN or in dotted format notation (for example 201.10.8.1). You can also specify the selected port in the format: <IP Address>:<port>. This parameter is applicable only if you select 'Yes' in the 'Is Proxy Used' field. If you enable Proxy Redundancy (by setting EnableProxyKeepAlive=1 or 2), the gateway can function with up to four Proxy servers. If there is no response from the primary Proxy, the gateway tries to communicate with the redundant Proxies. When a redundant Proxy is found, the gateway either continues working with it until the next failure occurs or reverts to the primary Proxy (refer to the 'Redundancy Mode' parameter). If none of the Proxy servers respond, the gateway goes over the list again. The gateway also provides real time switching (hotswap mode), between the primary and redundant proxies ('IsProxyHotSwap=1'). If the first Proxy doesn't respond to INVITE message, the same INVITE message is immediately sent to the second Proxy. **Note 1:** If 'EnableProxyKeepAlive=1 or 2', the gateway monitors the connection with the Proxies by using keep-alive messages (OPTIONS or REGISTER). **Note 2:** To use Proxy Redundancy, you must specify one or more redundant Proxies using multiple 'ProxyIP= <IP address>' definitions. **Note 3:** When port number is specified (e.g., domain.com:5080), DNS NAPTR/SRV queries aren't performed, even if ProxyDNSQueryType is set to 1 or 2.

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
ProxyIP **ProxyIP** **ProxyIP** [Redundant Proxy IP Address]	IP addresses of the redundant Proxies you are using. Enter the IP address as FQDN or in dotted format notation (for example 192.10.1.255). You can also specify the selected port in the format: <IP Address>:<port>. **Note 1:** This parameter is available only if you select 'Yes' in the 'Is Proxy Used' field. **Note 2:** When port number is specified, DNS NAPTR/SRV queries aren't performed, even if ProxyDNSQueryType is set to 1 or 2. **Note *ini* file:** The IP addresses of the redundant Proxies are defined by the second, third and fourth repetition of the *ini* file parameter 'ProxyIP'.
ProxyName [Proxy Name]	Home Proxy Domain Name. If specified, the name is used as Request-URI in REGISTER, INVITE and other SIP messages. If the proxy name isn't specified, the Proxy IP address is used instead.
EnableProxySRVQuery [Enable Proxy SRV Queries]	This parameter is now obsolete. Please use the parameter ProxyDNSQueryType.
EnableSRVQuery [Enable SRV Queries]	This parameter is now obsolete. Please use the parameter DNSQueryType.
AlwaysSendToProxy [Always Use Proxy]	0 = Use standard SIP routing rules (default). 1 = All SIP messages and Responses are sent to Proxy server. **Note:** Applicable only if Proxy server is used.
SendInviteToProxy [Send All INVITE to Proxy]	0 = INVITE messages, generated as a result of Transfer or Redirect, are sent directly to the URI (according to the Refer-To header in the REFER message or Contact header in 30x response) (default). 1 = All INVITE messages, including those generated as a result of Transfer or Redirect are sent to Proxy. **Note:** Applicable only if Proxy server is used and 'AlwaysSendtoProxy=0'.
PreferRouteTable [Prefer Routing Table]	Determines if the local Tel to IP routing table takes precedence over a Proxy for routing calls. 0 = Only Proxy is used to route calls (default). 1 = The gateway checks the 'Destination IP Address' field in the 'Tel to IP Routing' table for a match with the outgoing call. Only if a match is not found, a Proxy is used. **Note:** Applicable only if Proxy is not always used ('AlwaysSendToProxy' = 0, 'SendInviteToProxy' = 0).
EnableProxyKeepAlive [Enable Proxy Keep Alive]	0 = Disable (default). 1 = Enable Keep alive with Proxy using OPTIONS. 2 = Enable Keep alive with Proxy using REGISTER. If EnableProxyKeepAlive = 1, SIP OPTIONS message is sent every ProxyKeepAliveTime. If EnableProxyKeepAlive = 2, SIP REGISTER message is sent every RegistrationTime. Any response from the Proxy, either success (200 OK) or failure (4xx response) is considered as if the Proxy is correctly communicating. **Note 1:** This parameter must be set to 1 (OPTIONS) when Proxy redundancy is used. **Note 2:** When EnableProxyKeepAlive = 2 (REGISTER), the homing redundancy mode is disabled. **Note 3:** When the active proxy doesn't respond to INVITE messages sent by the gateway, the proxy is marked as offline. The behavior is similar to a Keep-Alive (OPTIONS or REGISTER) failure.
ProxyKeepAliveTime [Proxy Keep Alive Time]	Defines the Proxy keep-alive time interval (in seconds) between Keep-Alive messages. The default value is 60 seconds. **Note:** This parameter is applicable only if EnableProxyKeepAlive = 1 (OPTIONS). When EnableProxyKeepAlive = 2 (REGISTER), the time interval between Keep-Alive messages is determined by the parameter RegistrationTime.

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

ini File Field Name Web Parameter Name	Valid Range and Description
DNSQueryType [DNS Query Type]	Enables the use of DNS Naming Authority Pointer (NAPTR) and Service Record (SRV) queries to resolve Proxy and Registrar servers and to resolve all domain names that appear in the Contact and Record-Route headers. Valid options include: 0 = A-Record (default) 1 = SRV 2 = NAPTR If set to A-Record (0), no NAPTR or SRV queries are performed. If set to SRV (1), and the Proxy / Registrar IP address parameter, the domain name in the Contact / Record-Route headers, or the IP address defined in the Routing tables contains a domain name without port definition, an SRV query is performed. The gateway uses the first host name received from the SRV query. The gateway then performs a DNS A-record query for the host name to locate an IP address. If set to NAPTR (2), an NAPTR query is performed. If it is successful, an SRV query is sent according to the information received in the NAPTR response. If the NAPTR query fails, an SRV query is performed according to the configured transport type. If the Proxy / Registrar IP address parameter, the domain name in the Contact / Record-Route headers, or the IP address defined in the Routing tables contains a domain name with port definition, the gateway performs a regular DNS A-record query. **Note:** To enable NAPTR/SRV queries for Proxy servers only, use the parameter ProxyDNSQueryType.
ProxyDNSQueryType [Proxy DNS Query Type]	Enables the use of DNS Naming Authority Pointer (NAPTR) and Service Record (SRV) queries to discover Proxy servers. Valid options include: 0 = A-Record (default) 1 = SRV 2 = NAPTR If set to A-Record, no NAPTR or SRV queries are performed. If set to SRV and the Proxy IP address parameter contains a domain name without port definition (e.g., ProxyIP = domain.com), an SRV query is performed. The SRV query returns up to four Proxy host names and their weights. The gateway then performs DNS A-record queries for each Proxy host name (according to the received weights) to locate up to four Proxy IP addresses. Therefore, if the first SRV query returns two domain names, and the A-record queries return two IP addresses each, no more searches are performed. If set to NAPTR, an NAPTR query is performed. If it is successful, an SRV query is sent according to the information received in the NAPTR response. If the NAPTR query fails, an SRV query is performed according to the configured transport type. If the Proxy IP address parameter contains a domain name with port definition (e.g., ProxyIP = domain.com:5080), the gateway performs a regular DNS A-record query. **Note:** When enabled, NAPTR/SRV queries are used to discover Proxy servers even if the parameter DNSQueryType is disabled.

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
UseSIPTgrp [Use Tgrp Information]	0 = Tgrp parameter isn't used (default). 1 = (send only) The trunk group number is added as the 'tgrp' parameter to the Contact header of outgoing SIP messages. If a trunk group number is not associated with the call, the 'tgrp' parameter isn't included. If a 'tgrp' value is specified in incoming messages, it is ignored. 2 = (send and receive) The functionality of outgoing SIP messages is identical to the functionality described in option (1). In addition, for incoming SIP messages, if the Request-URI includes a 'tgrp' parameter, the gateway routes the call according to that value (if possible). If the Contact header includes a 'tgrp' parameter, it is copied to the corresponding outgoing messages in that dialog.
EnableGRUU [Enable GRUU]	Determines whether or not the GRUU mechanism is used. Valid options include: 0 = Disable (default) 1 = Enable The gateway obtains a GRUU by generating a normal REGISTER request. This request contains a Supported header field with the value "gruu". The gateway includes a "+sip.instance" Contact header field parameter for each contact for which the GRUU is desired. This Contact parameter contains a globally unique ID that identifies the gateway instance. The global unique id is as follows: ▪ If registration is per endpoint (AuthenticationMode=0), it is the MAC address of the gateway concatenated with the phone number of the endpoint. ▪ If the registration is per gateway (AuthenticationMode=1) it is only the MAC address. ▪ When the "User Information" mechanism is used, the globally unique ID is the MAC address concatenated with the phone number of the endpoint (defined in the User-Info file). If the Registrar/Proxy supports GRUU, the REGISTER responses contain the "gruu" parameter in each Contact header field. The Registrar/Proxy provides the same GRUU for the same AOR and instance-id in case of sending REGISTER again after expiration of the registration. The gateway places the GRUU in any header field which contains a URI. It uses the GRUU in the following messages: INVITE requests, 2xx responses to INVITE, SUBSCRIBE requests, 2xx responses to SUBSCRIBE, NOTIFY requests, REFER requests, and 2xx responses to REFER. **Note:** If the GRUU contains the "opaque" URI parameter, the gateway obtains the AOR for the user by stripping the parameter. The resulting URI is the AOR. For example: AOR: sip:alice@example.com GRUU: sip:alice@example.com;opaque="kjh29x97us97d"
UserAgentDisplayInfo [User-Agent Information]	Defines the string that is used in the SIP request header 'User-Agent' and SIP response header 'Server'. If not configured, the default string 'AudioCodes product-name s/w-version' is used (e.g., User-Agent: Audiocodes-Sip-Gateway-Mediant 2000/v.4.80.004.008). When configured, the string 'UserAgentDisplayInfo s/w-version' is used (e.g., User-Agent: MyNewOEM/v.4.80.004.008). Note that the version number can't be modified. The maximum string length is 50 characters.
SessionExpiresMethod [Session Expires Method]	Defines the SIP method used for session-timer updates. 0 = Use Re-INVITE messages for session-timer updates (default). 1 = Use UPDATE messages. **Note 1:** The gateway can receive session-timer refreshes using both methods. **Note 2:** The UPDATE message used for session-timer purposes is not included in the SDP body.

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
UseGatewayNameForOptions [Use Gateway Name for OPTIONS]	0 = Use the gateway's IP address in keep-alive OPTIONS messages (default). 1 = Use 'GatewayName' in keep-alive OPTIONS messages. The OPTIONS Request-URI host part contains either the gateway's IP address or a string defined by the parameter 'Gatewayname'. The gateway uses the OPTIONS request as a keep-alive message to its primary and redundant Proxies (EnableProxyKeepAlive = 1).
IsProxyHotSwap [Enable Proxy Hotswap]	Enable Proxy Hot Swap redundancy mode. 0 = Disabled (default) 1 = Enabled If Hot Swap is enabled, SIP INVITE message is first sent to the primary Proxy server. If there is no response from the primary Proxy server for 'ProxyHotSwapRtx' retransmissions, the INVITE message is resent to the redundant Proxy server.
ProxyHotSwapRtx [Number of RTX before Hotswap]	Number of retransmitted INVITE messages before call is routed (hot swap) to another Proxy The valid range is 1 to 30. The default value is 3. **Note:** This parameter is also used for alternative routing using the Tel to IP Routing table. If a domain name in the routing table is resolved into 2 IP addresses, and if there is no response for 'ProxyHotSwapRtx' retransmissions to the INVITE message that is sent to the first IP address, the gateway immediately initiates a call to the second IP address.
ProxyRedundancyMode [Redundancy Mode]	0 (Parking mode) = the gateway continues working with the last active Proxy until the next failure (default). 1(Homing mode) = the gateway always tries to work with the primary Proxy server (switches back to the primary Proxy whenever it is available). **Note:** To use ProxyRedundancyMode, enable Keep-alive with Proxy option (EnableProxyKeepAlive=1 or 2).

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

ini File Field Name Web Parameter Name	Valid Range and Description
ProxyLoadBalancingMethod [Proxy Load Balancing Method]	Enables the usage of the Proxy Load Balancing mechanism. Valid options include: 0 = Load Balancing is disabled (default). 1 = Round Robin. 2 = Random Weights. When Round Robin (1) algorithm is used, a list of all possible Proxy IP addresses is compiled. This list includes all entries in the ProxyIP table after necessary DNS resolutions (including NAPTR and SRV, if configured). This list can handle up to 15 entries. After this list is compiled, the Proxy Keep-Alive mechanism (according to EnableProxyKeepAlive and ProxyKeepAliveTime) is used to mark each entry as Offline or Online. The balancing is only performed on Proxy servers that are marked as Online. All outgoing messages are equally distributed across the Proxy IP list. REGISTER messages are also distributed unless a RegistrarIP is configured. The Proxy IP list is refreshed according to ProxyIPListRefreshTime. If a change in the order of the entries in the list occurs, all load statistics are erased and balancing starts over again. When Random Weights (2) algorithm is used, the outgoing requests are not distributed equally among the Proxies. The weights are received from the DNS server by using SRV records. The gateway sends the requests in such a fashion that each Proxy receives a percentage of the requests according to its assigned weight. Load Balancing is not used in the following scenarios: ▪ The ProxyIP table includes more than one entry. ▪ The only Proxy defined is an IP address and not an FQDN. ▪ SRV usage is not enabled (DNSQueryType). ▪ The SRV response includes several records with a different Priority value.
ProxyIPListRefreshTime [Proxy IP List Refresh Time]	Defines the time interval (in seconds) between refreshes of the Proxy IP list. This parameter is used only when ProxyLoadBalancingMethod = 1. The interval range is 5 to 2,000,000. The default interval is 60.
IsFallbackUsed [Enable Fallback to Routing Table]	0 = Gateway fallback is not used (default). 1 = Internal Tel to IP Routing table is used when Proxy servers are not available. When the gateway falls back to the internal Tel to IP Routing table, the gateway continues scanning for a Proxy. When the gateway locates an active Proxy, it switches from internal routing back to Proxy routing. **Note:** To enable the redundant Proxies mechanism set 'EnableProxyKeepAlive' to 1 or 2.
UserName [User Name]	Username used for Registration and for Basic/Digest authentication process with Proxy / Registrar. Parameter doesn't have a default value (empty string).
Password [Password]	Password used for Basic/Digest authentication process with Proxy / Registrar. The default is 'Default_Passwd'.
Cnonce [Cnonce]	String used by the SIP Server and client to provide mutual authentication (free format, i.e., 'Cnonce = 0a4f113b'). The default is 'Default_Cnonce'.
IsRegisterNeeded [Enable Registration]	0 = Gateway does not register to Proxy/Registrar (default). 1 = Gateway registers to Proxy/Registrar at power up. **Note:** The gateway sends a REGISTER request for each channel or for the entire gateway (according to the parameter AuthenticationMode).

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

ini **File Field Name** **Web Parameter Name**	**Valid Range and Description**
RegistrarIP [Registrar IP Address]	IP address and optionally port number of Registrar server. Enter the IP address in dotted format notation, for example 201.10.8.1:<5080>. **Note 1:** If not specified, the REGISTER request is sent to the primary Proxy server (refer to 'Proxy IP address' parameter). **Note 2:** When port number is specified, DNS NAPTR/SRV queries aren't performed, even if DNSQueryType is set to 1 or 2.
RegistrarName [Registrar Name]	Registrar domain name. If specified, the name is used as Request-URI in REGISTER messages. If isn't specified (default), the Registrar IP address or Proxy name or Proxy IP address is used instead.
GWRegistrationName [Gateway Registration Name]	Defines the user name that is used in From and To headers of REGISTER messages. If 'GWRegistrationName' isn't specified (default), the 'Username' parameter is used instead. **Note:** This parameter is applicable only to a single gateway registration (AuthenticationMode=1). When the gateway registers each B-channel separately (AuthenticationMode=0), the user name is set to the endpoint's phone number002E
AuthenticationMode [Authentication Mode]	0 (Per Endpoint) = Registration & Authentication separately for each B-channel. 1 (Per Gateway) = Single Registration & Authentication for the gateway (default).
RegistrationTime [Registration Time]	Defines the time (in seconds) for which registration to a Proxy server is valid. The value is used in the header 'Expires'. In addition, this parameter defines the time interval between Keep-Alive messages when EnableProxyKeepAlive = 2 (REGISTER). Typically, a value of 3600 should be assigned for one hour registration. The gateway resumes registration according to the parameter RegistrationTimeDivider. The default is 180 seconds.
RegistrationTimeDivider [Re-registration Timing (%)]	Defines the re-registration timing (in percentage). The timing is a percentage of the re-register timing set by the Registration server. The valid range is 50 to 100. The default value is 50. For example: If 'RegistrationTimeDivider = 70' (%) and Registration Expires time = 3600, the gateway resends its registration request after 3600 x 70% = 2520 sec.
RegistrationRetryTime [Registration Retry Time]	Defines the time period (in seconds) after which a Registration request is resent if registration fails with 4xx, or there is no response from the Proxy/Registrar. The default is 30 seconds. The range is 10 to 3600.
NumberOfActiveDialogs	Defines the maximum number of active SIP dialogs that are not call related (i.e., REGISTER and SUBSCRIBE). This parameter is used to control the Registration / Subscription rate. The valid range is 1 to 20. The default value is 20.
PrackMode [PRACK Mode]	PRACK mechanism mode for 1xx reliable responses: 0 = Disabled. 1 = Supported (default). 2 = Required. **Note 1:** The Supported and Required headers contain the '100rel' parameter. **Note 2:** The gateway sends PRACK message if 180/183 response is received with '100rel' in the Supported or the Required headers.

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

ini File Field Name Web Parameter Name	Valid Range and Description
AssertedIdMode [Asserted Identity Mode]	0 = None (default). 1 = P-asserted. 2 = P-preferred. The Asserted ID mode defines the header that is used in the generated INVITE request. The header also depends on the calling Privacy: allowed or restricted. The P-asserted (or P-preferred) headers are used to present the originating party's Caller ID. The Caller ID is composed of a Calling Number and (optionally) a Calling Name. P-asserted (or P-preferred) headers are used together with the Privacy header. If Caller ID is restricted, the 'Privacy: id' is included. Otherwise, for allowed Caller ID the 'Privacy: none' is used. If Caller ID (received from PSTN) is restricted, the From header is set to <anonymous@anonymous.invalid>.
UseTelURIForAssertedID [Use Tel URI for Asserted Identity]	Determines the format of the URI in the P-Asserted and P-Preferred headers. 0 = 'sip:' (default). 1 = 'tel:'.
EnableRPIheader [Enable Remote Party ID]	Enable Remote-Party-ID (RPI) headers for calling and called numbers for Tel→IP calls. 0 = Disabled (default). 1 = RPI headers are generated in SIP INVITE messages for both called and calling numbers.
IsUserPhone [Use "user=phone" in SIP URL]	0 = Doesn't use 'user=phone' string in SIP URI. 1 = 'user=phone' string is part of the SIP URI (default).
IsUserPhoneInFrom [Use "user=phone" in From header]	0 = Doesn't use ';user=phone' string in From header (default). 1 = ';user=phone' string is part of the From header.
IsUseToHeaderAsCalledNumber	0 = Sets the destination number to the user part of the Request-URI for IP→Tel calls, and sets the 'Contact' header to the source number for Tel→ IP calls (default). 1 = Sets the destination number to the user part of the 'To' header for IP→Tel calls, and sets the 'Contact' header to the *username* parameter for Tel→IP calls.

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

<table>
<tr><th>ini File Field Name
Web Parameter Name</th><th>Valid Range and Description</th></tr>
<tr><td>EnableHistoryInfo
[Enable History-Info Header]</td><td>Enables usage of the History-Info header.
Valid options include:
0 = Disable (default)
1 = Enable
<u>UAC Behavior:</u>

Initial request: The History-Info header is equal to the Request URI. If a PSTN Redirect number is received, it is added as an additional History-Info header with an appropriate reason.
Upon receiving the final failure response, the gateway copies the History-Info as is, adds the reason of the failure response to the last entry, and concatenates a new destination to it (if an additional request is sent).
The order of the reasons is as follows:
- Q.850 Reason
- SIP Reason
- SIP Response code
Upon receiving the final (success or failure) response, the gateway searches for a Redirect reason in the History-Info (i.e., 3xx/4xx SIP Reason). If found, it is passed to ISDN, according to the following table:

<table>
<tr><th>SIP Reason Code</th><th>ISDN Redirecting Reason</th></tr>
<tr><td>302 – Moved Temporarily</td><td>Call Forward Universal (CFU)</td></tr>
<tr><td>408 – Request Timeout</td><td rowspan="3">Call Forward No Answer (CFNA)</td></tr>
<tr><td>480 – Temporarily Unavailable</td></tr>
<tr><td>487 – Request Terminated</td></tr>
<tr><td>486 – Busy Here</td><td rowspan="2">Call Forward Busy (CFB)</td></tr>
<tr><td>600 – Busy Everywhere</td></tr>
</table>

If history reason is a Q.850 reason, it is translated to the SIP reason (according to the SIP-ISDN tables) and then to ISDN Redirect reason according to the table above.

<u>UAS Behavior:</u>

History-Info is sent in the final response only.
Upon receiving a request with History-Info, the UAS checks the policy in the request. If 'session', 'header', or 'history' policy tag is found, the (final) response is sent without History-Info. Otherwise, it is copied from the request.
</td></tr>
<tr><td>SIPSubject
[Subject]</td><td>Defines the value of the Subject header in outgoing INVITE messages. If not specified, the Subject header isn’t included (default).
The maximum length of the subject is limited to 50 characters.</td></tr>
<tr><td>MultiPtimeFormat
[Multiple Packetization Time Format]</td><td>Determines whether the 'mptime' attribute is included in the outgoing SDP.
Valid options include:
0 = Disable (default)
1 = Enable (includes the mptime attribute in the outgoing SDP -- PacketCable defined format)
The 'mptime' attribute enables the gateway to define a separate Packetization period for each negotiated coder in the SDP. The 'mptime' attribute is only included if this parameter is enabled, even if the remote side includes it in the SDP offer.
Upon reception, each coder receives its 'ptime' value in the following precedence:

From 'mptime' attribute.
From 'ptime' attribute.
Default value.
</td></tr>
</table>

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
EnableReasonHeader [Enable Reason Header]	Enables / disables the usage of the SIP Reason header. 0 = Disable. 1 = Enable (default).
EnablePtime	0 = Remove the ptime header from SDP. 1 = Include the ptime header in SDP (default).
EnableUserInfoUsage [Enable User-Information Usage]	Enables or disables usage of the User Information loaded to the gateway via the User Information auxiliary file. 0 = Disable (default). 1 = Enable.
CoderName	Defines the gateway's coder list (up to five coders can be configured). Enter coders in the following format: CoderName=<Coder Name>,<Ptime>,<Rate>,<Payload Type>,<Silence Suppression Mode>.

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711 A-law [g711Alaw64k]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	Always 64	Always 8	Disable [0] Enable [1]
G.711 μ-law [g711Ulaw64k]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	Always 64	Always 0	Disable [0] Enable [1]
G.729 [g729]	10, 20 (default), 30, 40, 50, 60, 80, 100	Always 8	Always 18	Disable [0] Enable [1] Enable w/o Adaptations [2]
G.723.1 [g7231]	30 (default), 60, 90, 120	5.3 [0], 6.3 [1] (default)	Always 4	Disable [0] Enable [1]
G.726 [g726]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	16 [0], 24 [1], 32 [2] (default), 40 [3]	Dynamic (0-120)	Disable [0] Enable [1]
GSM-FR [gsmFullRate]	20 (default), 40, 60, 80	Always 13	Always 3	Disable [0] Enable [1]
GSM-EFR [gsmEnhancedFullRate]	10, 20 (default), 30, 40, 50, 60, 80, 100	12.2	Dynamic (0-120)	Disable [0] Enable [1]
MS-GSM [gsmMS]	40 (default)	Always 13	Always 3	Disable [0] Enable [1]
NetCoder [NetCoder]	20 (default), 40, 60, 80, 100, 120	6.4 [0] 7.2 [1] 8.0 [2] 8.8 [3] (default)	51 52 53 54	Disable [0] Enable [1]
AMR [Amr]	20 (default)	4.75 [0], 5.15 [1], 5.90 [2], 6.70 [3], 7.40 [4], 7.95 [5], 10.2 [6], 12.2 [7] (default)	Dynamic (0-120)	Disable [0] Enable [1]
EVRC [Evrc]	20 (default), 40, 60, 80, 100	Variable [0] (default), 1/8 [1], 1/2 [3], Full [4]	Dynamic (0-120)	Disable [0] Enable [1]

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**				
	Transparent [Transparent]	20 (default), 40, 60, 80, 100, 120	Always 64	Dynamic (0-120)	Disable [0] Enable [1]
	QCELP [QCELP]	20 (default), 40, 60, 80, 100, 120	Always 13	Always 12	Disable [0] Enable [1]
	iLBC [iLBC	20 (default), 40, 60, 80, 100, 120	15 (default)	Dynamic (0-120)	Disable [0] Enable [1]
		30 (default), 60, 90, 120	13		
	T.38 [t38fax]	N/A	N/A	N/A	N/A
	G.711A-law_VBD [g711AlawVbd]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	Always 64	Dynamic (0-120)	N/A
	G.711U-law_VBD [g711UlawVbd]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	Always 64	Dynamic (0-120)	N/A
	Note 1: The coder name is case-sensitive. **Note 2:** If silence suppression is not defined (for a specific coder), the value defined by the parameter EnableSilenceCompression is used. **Note 3:** The value of several fields is hard-coded according to well-known standards (e.g., the payload type of G.711 U-law is always 0). Other values can be set dynamically. If no value is specified for a dynamic field, a default value is assigned. If a value is specified for a hard-coded field, the value is ignored. **Note 4:** Only the ptime of the first coder in the defined coder list is declared in INVITE / 200 OK SDP, even if multiple coders are defined. **Note 5:** If the coder G.729 is selected and silence suppression is disabled (for this coder), the gateway includes the string 'annexb=no' in the SDP of the relevant SIP messages. If silence suppression is enabled or set to 'Enable w/o Adaptations', 'annexb=yes' is included. An exception to this logic is when the remote gateway is a Cisco device (IsCiscoSCEMode). **Note 6:** Both GSM-FR and MS-GSM coders use Payload Type = 3. Using SDP, it isn't possible to differentiate between the two. Therefore, it is highly recommended not to select both coders simultaneously. For example: CoderName = g711Alaw64k,20,,,0 CoderName = g711Ulaw64k,40 CoderName = g7231,90,1,,1 CoderName = g726,$$,2,,0				
DSPVersionTemplateNumber [DSP Version Template Number]	Determines the number of the DSP load. Each load has a different coder list, a different channel capacity and different supported features. Available values are 0, 1, 2 and 3. For detailed information on the supported coders and channel capacity, refer to the Release Notes.				
VBRCoderHeaderFormat	Defines the format of the RTP header for VBR coders. 0 = Payload only (no header, no TOC, no m-factor) (default). 1 = Supports RFC 2658 - 1 byte for interleaving header (always 0), TOC, no m-factor. 2 = Payload including TOC only, allow m-factor. 3 = RFC 3558.				

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
TransparentCoderOnDataCall	0 = Only use coders from the coder list (default). 1 = Use transparent coder for data calls (according to RFC 4040). The 'Transparent' coder can be used on data calls. When the gateway receives a Setup message from the ISDN with 'TransferCapabilities = data', it can initiate a call using the coder 'Transparent' (even if the coder is not included in the coder list). The initiated INVITE includes the following SDP attribute: a=rtpmap:97 CLEARMODE/8000 The default Payload Type is set according to the CoderName table. If the Transparent coder is not set in the Coders table, the default value is set to 56. The Payload Type is negotiated with the remote side, i.e., the selected Payload Type is according to the remote side selection. The receiving gateway must include the 'Transparent' coder in its coder list.
IsFaxUsed [Fax Signaling Method]	Determines the SIP signaling method used to establish and convey a fax session after a fax is detected. 0 = No fax negotiation using SIP signaling (default). 1 = Initiates T.38 fax relay. 2 = Initiates fax using the coder G.711 A-law/μ-law with adaptations (refer to note 1). 3 = Initiates T.38 fax relay. If the T.38 negotiation fails, the gateway re-initiates a fax session using the coder G.711 A-law/μ-law with adaptations (see note 1). **Note 1:** Fax adaptations: Echo Canceller = On Silence Compression = Off Echo Canceller Non-Linear Processor Mode = Off Dynamic Jitter Buffer Minimum Delay = 40 Dynamic Jitter Buffer Optimization Factor = 13 **Note 2:** If the gateway initiates a fax session using G.711 (option 2 and possibly 3), a 'gpmd' attribute is added to the SDP in the following format: For A-law: 'a=gpmd:0 vbd=yes;ecan=on'. For μ-law: 'a=gpmd:8 vbd=yes;ecan=on'. **Note 3:** When 'IsFaxUsed' is set to 1, 2 or 3 the parameter 'FaxTransportMode' is ignored. **Note 4:** When the value of IsFaxUsed is other than 1, T.38 might still be used without the control protocol's involvement. To completely disable T.38, set FaxTransportMode to a value other than 1.
T38UseRTPPort	Defines that the T.38 packets are sent / received using the same port as RTP packets. 0 = Use the RTP port +2 to send / receive T.38 packets (default). 1 = Use the same port as the RTP port to send / receive T.38 packets.
CngDetectorMode [CNG Detector Mode]	0 = Disable (default). Don't detect CNG. 1 = Relay. CNG is detected on the originating side. CNG packets are sent to the remote side according to T.38 (if IsFaxUsed=1) and the fax session is started. 2 = Events Only. CNG is detected on the originating side. The CNG signal passes transparently to the remote side and fax session is started. Usually T.38 fax session starts when the 'preamble' signal is detected by the answering side. Some SIP gateways don't support the detection of this fax signal on the answering side, thus, for these cases it is possible to configure the gateways to start the T.38 fax session when the CNG tone is detected by the originating side. However, this mode is not recommended.

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
DefaultReleaseCause [Default Release Cause]	Default Release Cause (for IP to Tel calls), used when the gateway initiates a call release, and if an explicit matching cause for this release isn't found, a default release cause can be configured. The default release cause is described in the Q.931 notation, and translated to corresponding SIP equivalent response value The default release cause is: NO_ROUTE_TO_DESTINATION (3). Other common values are: NO_CIRCUIT_AVAILABLE (34) or DESTINATION_OUT_OF_ORDER (27), etc. **Note:** The default release cause is described in the Q.931 notation, and is translated to corresponding SIP 40x or 50x value. For example: 404 for 3, 503 for 34 and 502 for 27. For mapping of SIP to Q.931 and Q.931 to SIP release causes, refer to Appendix H on page 379.
IPAlertTimeout [Tel to IP No Answer Timeout]	Defines the time (in seconds) the gateway waits for a 200 OK response from the called party (IP side) after sending an INVITE message. If the timer expires, the call is released. The valid range is 0 to 3600. The default value is 180.
SipSessionExpires [Session-Expires Time]	Determines the timeout (in seconds) for keeping a re-INVITE message alive within a SIP session. The SIP session is refreshed (using INVITE) each time this timer expires. The default is 0 (not activated).
MINSE [Minimum Session-Expires]	Defines the time (in seconds) that is used in the Min-SE header. This header defines the minimum time that the user agent supports for session refresh. The valid range is 10 to 100000. The default value is 90.
SIPMaxRtx [SIP Maximum Rtx]	Number of UDP transmissions (first transmission + retransmissions) of SIP messages. The range is 1 to 7. The default value is 7.
SipT1Rtx [SIP T1 Retransmission Timer (msec)]	The time interval (in msec) between the first transmission of a SIP message and the first retransmission of the same message. The default is 500. **Note:** The time interval between subsequent retransmissions of the same SIP message starts with SipT1Rtx and is multiplied by two until SipT2Rtx. For example (assuming that SipT1Rtx = 500 and SipT2Rtx = 4000): The first retransmission is sent after 500 msec. The second retransmission is sent after 1000 (2*500) msec. The third retransmission is sent after 2000 (2*1000) msec. The fourth retransmission and subsequent retransmissions until SIPMaxRtx are sent after 4000 (2*2000) msec.
SipT2Rtx [SIP T2 Retransmission Timer (msec)]	The maximum interval (in msec) between retransmission of SIP messages. The default is 4000. **Note:** The time interval between subsequent retransmissions of the same SIP message starts with SipT1Rtx and is multiplied by two until SipT2Rtx.
EnableEarlyMedia [Enable Early Media]	0 = Early Media is disabled (default). 1 = Enable Early Media. If enabled, the gateway gateway sends 183 Session Progress response with SDP (instead of 180 ringing), enabling the setup of the media stream prior to the answering of the call. Sending 183 response depends on the Progress Indicator. It is sent only if PI=1 or PI=8 was received in Proceeding or Alert PRI messages. For CAS gateways see the 'ProgressIndicator2IP' parameter. **Note:** Generally, this parameter is set to 1.

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
EnableTransfer [Enable Transfer]	0 = Call transfer is not allowed. 1 = The gateway responds to a REFER message with 'Referred-To' header to initiates a call transfer (default). **Note 1:** To use this service, the gateways at both ends must support this option. **Note 2:** To use this service, set the parameter 'Enable Hold' to 'Yes'.
XferPrefix [Transfer Prefix]	Defined string that is added, as a prefix, to the transferred called number, when REFER/3xx message is received. **Note 1:** The number manipulation rules apply to the user part of the Refer-To/Contact URI before it is sent in the INVITE message. **Note 2:** The xferprefix parameter can be used to apply different manipulation rules to differentiate the transferred number from the original dialed number.
EnableHold [Enable Hold]	0 = Hold service is disabled. 1 = Hold service is enabled, held tone is played to holding party (default).
EnableForward [Enable Call Forward]	0 = Disable call forward. 1 = Enable call forward service (default). The gateway doesn't initiate call forward, it can only respond to call forward requests.
EnableCallWaiting [Enable Call Waiting]	0 = Disabled. 1 = Enabled (default). If enabled, when the gateway initiates a Tel to IP call to a destination that is busy, it plays a Call Waiting Ringback tone to the originator of the call. **Note 1:** The gateway's Call Progress Tones file must include a Call Waiting Ringback tone. **Note 2:** The EnableHold parameter must be enabled on the called side.
Send180ForCallWaiting	0 = Use 182 Queued response to indicate a call waiting (default). 1 = Use 180 Ringing response to indicate a call waiting.
HookFlashCode [Hook-Flash Code]	Determines the digit pattern used by the PBX to indicate a 'Hook-Flash' event. When this pattern is detected from the Tel side, the gateway responds as if a Hook-Flash event occurs and sends an INFO message indicating 'Hook Flash'. If configured and a Hook-Flash indication is received from the IP side, the gateway generates this pattern to the Tel side. The valid range is a 25-character string.
UseSIPURIForDiversionHeader	Sets the URI format in the Diversion header. 0 = 'tel:' (default). 1 = 'sip:'.
RxDTMFOption [Declare RFC 2833 in SDP]	Defines the supported Receive DTMF negotiation method. 0 = Don't declare RFC 2833 telephony-event parameter in SDP. 1 = N/A. 2 = N/A. 3 = Declare RFC 2833 'telephony-event' parameter in SDP (default). The gateway is designed to always be receptive to RFC 2833 DTMF relay packets. Therefore, it is always correct to include the 'telephony-event' parameter as a default in the SDP. However some gateways use the absence of the 'telephony-event' from the SDP to decide to send DTMF digits inband using G.711 coder, if this is the case you can set 'RxDTMFOption=0'.

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

ini **File Field Name** **Web Parameter Name**	**Valid Range and Description**
TxDTMFOption [DTMF RFC2833 Negotiation]	Determines a single or several preferred transmit DTMF negotiation methods. 0 (Not Supported) = No negotiation, DTMF digits are sent according to the parameters 'DTMFTransportType' and 'RFC2833PayloadType' (default). 1 (INFO Nortel) = Sends DTMF digits according to IETF <draft-choudhuri-sip-info-digit-00>. 2 (NOTIFY) = Sends DTMF digits according to <draft-mahy-sipping-signaled-digits-01>. 3 (INFO Cisco) = Sends DTMF digits according to Cisco format. 4 (RFC 2833). **Note 1:** DTMF negotiation methods are prioritized according to the order of their appearance. **Note 2:** When out-of-band DTMF transfer is used (1, 2 or 3), the parameter 'DTMFTransportType' is automatically set to 0 (DTMF digits are erased from the RTP stream). **Note 3:** When RFC 2833 (4) is selected, the gateway: • Negotiates RFC 2833 Payload Type (PT) using local and remote SDPs. • Sends DTMF packets using RFC 2833 PT according to the PT in the received SDP. • Expects to receive RFC 2833 packets with the same PT as configured by the parameter 'RFC2833PayloadType'. • Uses the same PT for send and receive if the remote party doesn't include the RFC 2833 DTMF PT in its SDP. **Note 4:** When TxDTMFOption is set to 0, the RFC 2833 PT is set according to the parameter 'RFC2833PayloadType' for both transmit and receive. ***ini*** **file note:** The DTMF transmit methods are defined using a repetition of the same (TxDTMFOption) parameter (up to five options can be provided).
DisableAutoDTMFMute	Enables / disables the automatic mute of DTMF digits when out-of-band DTMF transmission is used. 0 = Auto mute is used (default). 1 = No automatic mute of in-band DTMF. When 'DisableAutoDTMFMute=1', the DTMF transport type is set according to the parameter 'DTMFTransportType' and the DTMF digits aren't muted if out-of-band DTMF mode is selected ('TxDTMFOption =1, 2 or 3'). This enables the sending of DTMF digits in-band (transparent of RFC 2833) in addition to out-of-band DTMF messages. **Note:** Usually this mode is not recommended.
EnableReasonHeader [Enable Reason Header]	Enables or disables the usage of the SIP Reason header. Valid options include: 0 = Disable 1 = Enable (default)
EnableSemiAttendedTransfer [Enable Semi-Attended Transfer]	Determines the gateway behavior when a Transfer is initiated while still in Alerting state. Valid options include: 0 = Send REFER with Replaces (default). 1 = Send CANCEL, and after a 487 response is received, send REFER without Replaces.
3xxBehavior [3xx Behavior]	Determines the gateway's behavior when a 3xx response is received for an outgoing INVITE request. The gateway can either use the same call identifiers (CallID, branch, to and from tags) or change them in the new initiated INVITE. 0 (forward) = Use different call identifiers for a redirected INVITE message (default). 1 (redirect) = Use the same call identifiers.

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
MaxActiveCalls [Max Number Of Active Calls]	Defines the maximum number of calls that the gateway can have active at the same time. If the maximum number of calls is reached, new calls are not established. The default value is max available channels (no restriction on the maximum number of calls). The valid range is 1 to 240.
MaxCallDuration [Max Call Duration]	Defines the maximum call duration in minutes. If this time expires, both sides of the call are released (IP and Tel). The valid range is 0 to 120. The default is 0 (no limitation).
EnableBusyOut [Enable Busy Out]	0 = Not used (default) 1 = Enable busy out If Busy out is enabled, all E1/T1 trunks are automatically put out of service by sending a remote alarm (AIS) or Service Out message for T1 PRI trunks that support these messages (NI-2, 4/5-ESS, DMS-100 and Meridian), due to one of the following scenarios: • Physically disconnected from the network (i.e., Ethernet cable is disconnected). • The Ethernet cable is connected, but the gateway can't communicate with any host. Note that LAN Watch-Dog must be activated (EnableLANWatchDog = 1). • The gateway can't communicate with the gatekeeper/proxy (according to the Proxy keep-alive mechanism) and no other alternative exists to send the call. **Note:** The Busy out behavior varies between different protocol types.
EnableDigitDelivery2IP [Enable Digit Delivery to IP]	0 = Disabled (default). 1 = Enable digit delivery to IP. The digit delivery feature enables sending of DTMF digits to the destination IP address after the Tel→IP call was answered. To enable this feature, modify the called number to include at least one 'p' character. The gateway uses the digits before the 'p' character in the initial INVITE message. After the call was answered the gateway waits for the required time (# of 'p' * 1.5 seconds) and then sends the rest of the DTMF digits using the method chosen (in-band, out-of-band). **Note**: The called number can include several 'p' characters (1.5 seconds pause). For example, the called number can be as follows: 1001pp699, 8888p9p300.
EnableDigitDelivery [Enable Digit Delivery to Tel]	The digit delivery feature enables sending of DTMF digits to the gateway's B-channel after the call is answered. 0 = Disabled (default). 1 = Enable Digit Delivery feature for gateway (two stage dialing). **Note:** For incoming IP→Tel calls, if the called number includes the characters 'w' or 'p', the gateway places a call with the first part of the called number, and plays DTMF digits after the call is answered. If the character 'p' (pause) is used, the gateway waits for 1.5 seconds before playing the next DTMF digit. If the character 'w' is used, the gateway waits for detection of dial tone before it starts playing DTMF digits. The character 'w' can appear once in the called number, and must precede any 'p' character. The 'p' character can appear several times. For example: if the number '1007766p100' is defined as the called number, the gateway places a call with 1007766 as the destination number, then, after the call is answered, it waits for 1.5 seconds and plays the rest of the number (100) as DTMF digits. Other examples: 1664wpp102, 66644ppp503, 7774w100pp200.

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

ini **File Field Name** **Web Parameter Name**	**Valid Range and Description**
SITDetectorEnable	Enables or disables Special Information Tone (SIT) detection according to the ITU-T recommendation E.180/Q.35. Valid options include: 0 = Disable (default). 1 = Enable.
AMDTimeout	Timeout (in msec) between receiving CONNECT messages from the ISDN and sending of Answering Machine Detection (AMD) results. The valid range is 1 to 30,000. The default is 2,000 (2 seconds).
RTPOnlyMode	Enables the gateway to start sending and/or receiving RTP packets to and from remote endpoints without the need to establish a Control session. The remote IP address is determined according to the Tel2IP Routing table. The port is the same port as the local RTP port (set by BaseUDPPort and the channel on which the call was received). Valid options include: 0 = Disable (default). 1 = Transmit & Receive. 2 = Transmit Only. 3 = Receive Only.

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
Profile Parameters	
CoderName_ID	Defines groups of coders that can be associated with IP or Tel profiles (up to five coders in each group). Enter coder groups in the following format: CoderName_<coder group ID from 1 to 4>=<Coder Name>,<Ptime>,<Rate>,<Payload Type>,<Silence Suppression Mode>.

Coder Name	**Packetization Time**	**Rate**	**Payload Type**	**Silence Suppression**
G.711 A-law [g711Alaw64k]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	Always 64	Always 8	Disable [0] Enable [1]
G.711 μ-law [g711Ulaw64k]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	Always 64	Always 0	Disable [0] Enable [1]
G.729 [g729]	10, 20 (default), 30, 40, 50, 60, 80, 100	Always 8	Always 18	Disable [0] Enable [1] Enable w/o Adaptations [2]
G.723.1 [g7231]	30 (default), 60, 90, 120	5.3 [0], 6.3 [1] (default)	Always 4	Disable [0] Enable [1]
G.726 [g726]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	16 [0], 24 [1], 32 [2] (default), 40 [3]	Dynamic (0-120)	Disable [0] Enable [1]
GSM-FR [gsmFullRate]	20 (default), 40, 60, 80	Always 13	Always 3	Disable [0] Enable [1]
GSM-EFR [gsmEnhancedFullRate]	10, 20 (default), 30, 40, 50, 60, 80, 100	12.2	Dynamic (0-120)	Disable [0] Enable [1]
MS-GSM [gsmMS]	40 (default)	Always 13	Always 3	Disable [0] Enable [1]
NetCoder [NetCoder]	20 (default), 40, 60, 80, 100, 120	6.4 [0] 7.2 [1] 8.0 [2] 8.8 [3] (default)	51 52 53 54	Disable [0] Enable [1]
AMR [Amr]	20 (default)	4.75 [0], 5.15 [1], 5.90 [2], 6.70 [3], 7.40 [4], 7.95 [5], 10.2 [6], 12.2 [7] (default)	Dynamic (0-120)	Disable [0] Enable [1]
EVRC [Evrc]	20 (default), 40, 60, 80, 100	Variable [0] (default), 1/8 [1], 1/2 [3], Full [4]	Dynamic (0-120)	Disable [0] Enable [1]
Transparent [Transparent]	20 (default), 40, 60, 80, 100, 120	Always 64	Dynamic (0-120)	Disable [0] Enable [1]
QCELP [QCELP]	20 (default), 40, 60, 80, 100, 120	Always 13	Always 12	Disable [0] Enable [1]
iLBC [iLBC]	20 (default), 40, 60, 80, 100, 120	15 (default)	Dynamic (0-120)	Disable [0] Enable [1]

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

ini File Field Name Web Parameter Name	Valid Range and Description				
		30 (default), 60, 90, 120	13		
	T.38 [t38fax]	N/A	N/A	N/A	N/A
	G.711A-law_VBD [g711AlawVbd]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	Always 64	Dynamic (0-120)	N/A
	G.711U-law_VBD [g711UlawVbd]	10, 20 (default), 30, 40, 50, 60, 80, 100, 120	Always 64	Dynamic (0-120)	N/A
	Note 1: The coder name is case-sensitive. **Note 2:** If silence suppression is not defined (for a specific coder), the value defined by the parameter EnableSilenceCompression is used. **Note 3:** The value of several fields is hard-coded according to well-known standards (e.g., the payload type of G.711 U-law is always 0). Other values can be set dynamically. If no value is specified for a dynamic field, a default value is assigned. If a value is specified for a hard-coded field, the value is ignored. **Note 4:** Only the ptime of the first coder in the defined coder list is declared in INVITE / 200 OK SDP, even if multiple coders are defined. **Note 5:** If the coder G.729 is selected and silence suppression is enabled (for this coder), the gateway includes the string 'annexb=no' in the SDP of the relevant SIP messages. If silence suppression is set to 'Enable w/o Adaptations', 'annexb=yes' is included. An exception to this logic is when the remote gateway is a Cisco device (IsCiscoSCEMode). **Note 6:** Both GSM-FR and MS-GSM coders use Payload Type = 3. Using SDP, it isn't possible to differentiate between the two. Therefore, it is highly recommended not to select both coders simultaneously. **Note 7:** This parameter (CoderName_ID) can appear up to 20 times (five coders in four coder groups). For example: CoderName_1 = g711Alaw64k,20,,,0 CoderName_1 = g711Ulaw64k,40 CoderName_1 = g7231,90,1,,1 CoderName_2 = g726,$$,2,,0				

Table 6-7: SIP Configuration Parameters (continues on pages 150 to 169)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
IPProfile_ID [IP Profile Settings]	IPProfile_<Profile ID> = <Profile Name>,<Preference>,<Coder Group ID>,<IsFaxUsed*>, <DJBufMinDelay *>, <DJBufOptFactor *>,<IPDiffServ *>,<ControlIPDiffServ*>, <N/A use $$ instead>, <RTPRedundancyDepth>,<RemoteBaseUDPPort>,<CNGmode>, <VxxTransportType>, <NSEMode>,<N/A use $$ instead>, <PlayRBTone2IP>, <EnableEarlyMedia*>, <ProgressIndicator2IP*> Preference = (1-20) The preference option is used to determine the priority of the Profile. Where '20' is the highest preference value. If both IP and Tel profiles apply to the same call, the coders and other common parameters (noted by an asterisk) of the preferred Profile are applied to that call. If the Preference of the Tel and IP Profiles is identical, the Tel Profile parameters are applied. For example: IPProfile_1 = name1,2,1,0,10,13,15,44,1,1,6000,0,2,0,0,0,1,0 IPProfile_2 = name2,$$,$$,$$,$$,$$,$$,$$,$$,1,$$,$$,$$,$$,$$,$$,$$,$$ $$ = Not configured, the default value of the parameter is used. (*) = Common parameter used in both IP and Tel profiles. **Note 1:** The IP ProfileID can be used in the Tel2IP and IP2Tel routing tables (Prefix and PSTNPrefix parameters). **Note 2:** 'Profile Name' assigned to a ProfileID, enabling user to identify it intuitively and easily. **Note 3:** This parameter can appear up to 9 times (ID = 1 to 9).
TelProfile_ID [Tel Profile Settings]	TelProfile_<Profile ID> = <Profile Name>,<Preference>,<Coder Group ID>,<IsFaxUsed *>,<DJBufMinDelay *>, <DJBufOptFactor *>,<IPDiffServ *>,<ControlIPDiffServ*>,<DTMFVolume>,<InputGain>, <VoiceVolume>,<N/A use $$ instead>,<N/A use $$ instead>, <EnableDigitDelivery>, <ECE>, <N/A use $$ instead>, <N/A use $$ instead>, <FlashHookPeriod>, <EnableEarlyMedia>, <ProgressIndicator2IP> Preference = (1-10) The preference option is used to determine the priority of the Profile. Where '20' is the highest preference value. If both IP and Tel profiles apply to the same call, the coders and other common parameters (noted by an asterisk) of the preferred Profile are applied to that call. If the Preference of the Tel and IP Profiles is identical, the Tel Profile parameters are applied. For examples: TelProfile_1 = FaxProfile,1,1,1,40,13,22,33,$$,$$,$$,0,0,0,1,0,0,$$,0,$$ TelProfile_2 = ModemProfile,2,2,0,40,13,$$,$$,$$,$$,$$,$$,$$,$$,0,0,0,$$,0,$$ $$ = Not configured, the default value of the parameter is used. (*) = Common parameter used in both IP and Tel profiles. **Note 1:** The Tel ProfileID can be used in the Trunk Group table (TrunkGroup_x parameter). **Note 2:** 'Profile Name' assigned to a ProfileID, enabling users to identify it intuitively and easily. **Note 3:** This parameter can appear up to 4 times (ID = 1 to 4).

6.13 Voice Mail Parameters

For detailed information on the Voice Mail (VM) application, refer to the CPE Configuration Guide for Voice Mail.

Table 6-8: Voice Mail Configuration Parameters (continues on pages 170 to 171)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
VoiceMailInterface [Voice Mail Interface]	Enables the VM application on the gateway and determines the communication method used between the PBX and the gateway. 0 = None (default) 1 = DTMF 2 = SMDI 3 = QSIG 4 = SETUP Only (ISDN)
SubscriptionMode [Subscription Mode]	Determines the method the gateway uses to subscribe to an MWI server. Per Endpoint [0] = Each endpoint subscribes separately (default). Per Gateway [1] = Single subscription for the entire gateway.
SMDI [Enable SMDI]	Enables the Simplified Message Desk Interface (SMDI) on the gateway. 0 = Normal serial (default). 1 = Enable RS-232 SMDI interface. **Note:** When the RS-232 connection is used for SMDI messages (Serial SMDI) it cannot be used for other applications, for example, to access the Command Line Interface.
SMDITimeOut [SMDI Timeout]	Determines the time (in msec) that the gateway waits for an SMDI Call Status message before or after a Setup message is received. This parameter is used to synchronize the SMDI and CAS interfaces. If the timeout expires and only an SMDI message was received, the SMDI message is dropped. If the timeout expires and only a Setup message was received, the call is established. The valid range is 0 to 10000 (10 seconds). The default value is 2000.
LineTransferMode [Line Transfer Mode]	Determines the transfer method used by the gateway. 0 = IP (default). 1 = PBX blind transfer.
WaitForDialTime [Wait For Dial Time]	Determines the delay after hook-flash is generated and dialing is begun. Applies to call transfer (TrunkTransferMode = 3) on CAS gateways. The valid range (in milliseconds) is 0 to 20000 (20 seconds). The default value is 1000 (1 second).
MWIOnCode [MWI On Digit Pattern]	Determines a digit code used by the gateway to notify the PBX of messages waiting for a specific extension. This code is added as prefix to the dialed number. The valid range is a 25-character string.
MWIOffCode [MWI Off Digit Pattern]	Determines a digit code used by the gateway to notify the PBX that there aren't any messages waiting for a specific extension. This code is added as prefix to the dialed number. The valid range is a 25-character string.
TelDisconnectCode [Disconnect Call Digit Pattern]	Determines a digit pattern that, when received from the Tel side, indicates the gateway to disconnect the call. The valid range is a 25-character string.
The following digit pattern parameters apply only to VM applications that use the DTMF communication method. For the available patterns' syntaxes, refer to the User's Manual.	
DigitPatternForwardOnBusy [Forward on Busy Digit Pattern]	Determines the digit pattern used by the PBX to indicate 'call forward on busy'. The valid range is a 120-character string.
DigitPatternForwardOnNoAnswer [Forward on No Answer Digit Pattern]	Determines the digit pattern used by the PBX to indicate 'call forward on no answer'. The valid range is a 120-character string.

Table 6-8: Voice Mail Configuration Parameters (continues on pages 170 to 171)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
DigitPatternForwardOnDND [Forward on Do Not Disturb Digit Pattern]	Determines the digit pattern used by the PBX to indicate 'call forward on do not disturb'. The valid range is a 120-character string.
DigitPatternForwardNoReason [Forward on No Reason Digit Pattern]	Determines the digit pattern used by the PBX to indicate 'call forward with no reason'. The valid range is a 120-character string.
DigitPatternInternalCall [Internal Call Digit Pattern]	Determines the digit pattern used by the PBX to indicate an internal call. The valid range is a 120-character string.
DigitPatternExternalCall [External Call Digit Pattern]	Determines the digit pattern used by the PBX to indicate an external call. The valid range is a 120-character string.
Serial parameters (applicable only to the SMDI application).	
SerialBaudRate	Determines the value of the RS-232 baud rate. The valid range is: any value. It is recommended to use the following standard values: 1200, 2400, 9600 (default), 14400, 19200, 38400, 57600, 115200.
SerialData	Determines the value of the RS-232 data bit. 7 = 7-bit. 8 = 8-bit (default).
SerialParity	Determines the value of the RS-232 polarity. 0 = None (default). 1 = Odd. 2 = Even.
SerialStop	Determines the value of the RS-232 stop bit. 1 = 1-bit (default). 2 = 2-bit.
SerialFlowControl	Determines the value of the RS-232 flow control. 0 = None (default). 1 = Hardware.

6.14 ISDN and CAS Interworking-Related Parameters

Table 6-9: ISDN and CAS Interworking-Related Parameters (continues on pages 172 to 179)

ini **File Field Name** **Web Parameter Name**	**Valid Range and Description**
EnableTDMoverIP [Enable TDM Tunneling]	0 = Disabled (default). 1 = TDM Tunneling is enabled. When TDM Tunneling is enabled, the originating gateway automatically initiates SIP calls from all enabled B-channels belonging to the E1/T1/J1 spans that are configured with the 'Transparent' protocol. The called number of each call is the internal phone number of the B-channel that the call originates from. The IP to Trunk Group routing table is used to define the destination IP address of the terminating gateway. The terminating gateway automatically answers these calls if its E1/T1 protocol is set to 'Transparent' (ProtocolType = 5).
EnableISDNTunnelingTel2IP [Enable ISDN Tunneling Tel2IP]	Valid options include: 0 = Disable (default). 1 = Enable ISDN Tunneling from ISDN PRI to SIP using a proprietary SIP header 2 = Enable ISDN Tunneling from ISDN PRI to SIP using a dedicated message body When ISDN Tunneling is enabled, the gateway sends all ISDN PRI messages using the correlated SIP messages. Setup is tunneled using INVITE, all mid-call messages are tunneled using INFO, and Disconnect/Release is tunneled using BYE. The raw data from the ISDN is inserted into a proprietary SIP header (X-ISDNTunnelingInfo) or a dedicated message body (application/isdn) in the SIP messages. **Note:** It is necessary to set the parameter ISDNDuplicateQ931BuffMode to 128 (duplicate all messages) for this mechanism to function.
EnableISDNTunnelingIP2Tel [Enable ISDN Tunneling IP2Tel]	Valid options include: 0 = Disable (default) 1 = Enable ISDN Tunneling from SIP to ISDN PRI using a proprietary SIP header 2 = Enable ISDN Tunneling from SIP to ISDN PRI using a dedicated message body When ISDN Tunneling is enabled, the gateway extracts the raw data received in a proprietary SIP header (X-ISDNTunnelingInfo) or a dedicated message body (application/isdn) in the SIP messages and sends the data as ISDN messages to the PSTN side.
ISDNDuplicateQ931BuffMode	Controls the activation / deactivation of delivering raw Q.931 messages. 0 = ISDN messages aren't duplicated (default). 128 = All ISDN messages are duplicated. **Note:** This parameter is not updated on-the-fly and requires a gateway reset.
EnableQSIGTunneling [Enable QSIG Tunneling]	Enables QSIG tunneling over SIP according to <draft-elwell-sipping-qsig-tunnel-03>. 0 = Disable (default). 1 = Enable QSIG tunneling from QSIG to SIP and vice versa. When QSIG tunneling is enabled, all QSIG messages are sent as raw data in corresponding SIP messages using a dedicated message body. Note that QSIG tunneling must be enabled on both the originating and terminating gateways. **Note:** It is necessary to set the parameter 'ISDNDuplicateQ931BuffMode' to 128 (duplicate all messages) so that this mechanism can function.

Table 6-9: ISDN and CAS Interworking-Related Parameters (continues on pages 172 to 179)

ini File Field Name Web Parameter Name	Valid Range and Description
PlayRBTone2Trunk_ID [Play Ringback Tone to Trunk]	ID = Trunk number (0-7). 0 (Don't play) = The ISDN / CAS gateway doesn't play a Ringback Tone (RBT). No PI is sent to the ISDN, unless the parameter 'Progress Indicator to ISDN' is configured differently. 1 (Play) = The CAS gateway plays a local RBT to PSTN after receipt of a 180 ringing response (with or without SDP). **Note:** Reception of a 183 response doesn't cause the CAS gateway to play an RBT (unless 'SIP183Behaviour = 1'). The ISDN gateway functions according to the parameter 'LocalISDNRBSource': • If the ISDN gateway receives a 180 ringing response (with or without SDP) and 'LocalISDNRBSource = 1', it plays a RBT and sends an Alert with PI = 8 (unless the parameter 'Progress Indicator to ISDN' is configured differently). • If 'LocalISDNRBSource = 0', the ISDN gateway doesn't play an RBT and an Alert message (without PI) is sent to the ISDN. In this case, the PBX / PSTN should play the RBT to the originating terminal by itself. **Note:** Reception of a 183 response doesn't cause the ISDN gateway to play an RBT; the gateway issues a Progress message (unless 'SIP183Behaviour = 1'). If 'SIP183Behaviour = 1', the 183 response is treated the same way as a 180 ringing response. 2 = Play according to 'early media' (default). If a 180 response is received and the voice channel is already open (due to a previous 183 early media response or due to an SDP in the current 180 response), the ISDN / CAS gateway doesn't play the RBT; PI = 8 is sent in an ISDN Alert message (unless the parameter 'Progress Indicator to ISDN' is configured differently). If a 180 response is received but the 'early media' voice channel is not opened, the CAS gateway plays an RBT to the PSTN; the ISDN gateway functions according to the parameter 'LocalISDNRBSource': • If 'LocalISDNRBSource = 1', the ISDN gateway plays an RBT and sends an ISDN Alert with PI = 8 to the ISDN (unless the parameter 'Progress Indicator to ISDN' is configured differently). • If 'LocalISDNRBSource = 0', the ISDN gateway doesn't play an RBT. No PI is sent in the ISDN Alert message (unless the parameter 'Progress Indicator to ISDN' is configured differently). In this case, the PBX / PSTN should play an RBT tone to the originating terminal by itself. **Note:** Reception of a 183 response results in an ISDN Progress message (unless 'SIP183Behaviour = 1'). If 'SIP183Behaviour = 1' (183 is handled in the same way as a 180+SDP), the gateway sends an Alert message with PI = 8, without playing an RBT.
PlayRBTone2Tel [Play Ringback Tone to Tel]	Determines the method used to play Ringback tone to the Tel side. It applies to all trunks that are not configured by the parameter PlayRBTone2Trunk. Similar description as the parameter 'PlayRBTone2Trunk'.

Table 6-9: ISDN and CAS Interworking-Related Parameters (continues on pages 172 to 179)

ini File Field Name Web Parameter Name	Valid Range and Description
PlayRBTone2IP [Play Ringback Tone to IP]	0 = Ringback tone isn't played (default). 1 = Ringback tone is played (to IP) after SIP 183+SDP or 180+SDP response is sent. If configured to 1 ('Play'), and if EnableEarlyMedia = 1, for IP-to-Tel calls the gateway may play a ringback tone to IP, according to the following: • For CAS interfaces, the gateway opens a voice channel, sends a 183+SDP response and plays a Ringback tone to IP. • For ISDN interfaces, if a Progress or an Alert message with PI (1 or 8) is received from the ISDN, the gateway opens a voice channel, sends a 183+SDP or 180+SDP response, but it doesn't play a Ringback tone to IP. If PI (1 or 8) is received from the ISDN, the gateway assumes that Ringback tone is played by the ISDN Switch. Otherwise, the fateway plays a Ringback tone to IP after receiving an Alert message from the ISDN. It sends a 180+SDP response, signaling to the originating party to open a voice channel to hear the played Ringback tone. **Note 1:** To enable the gateway to send a 183/180+SDP responses, set EnableEarlyMedia to 1. **Note 2:** If EnableDigitDelivery = 1, the gateway doesn't play a Ringback tone to IP and doesn't send 183 or 180+SDP responses.
DefaultCauseMapISDN2IP [Default Cause Mapping From ISDN to IP]	Defines a single default ISDN Release Cause that is used (in ISDN to IP calls) instead of all received release causes except when the following Q.931 cause values are received: Normal Call Clearing (16), User Busy (17), No User Responding (18) or No Answer from User (19). The range is valid Q.931 release causes (0 to 127). The default value is 0 (indicates that the parameter is not configured - static mapping is used).
CauseMapSIP2ISDN_ID [Release Cause Mapping from SIP to ISDN]	Defines a flexible mapping of SIP Responses and Q.850 Release Causes. CauseMapSIP2ISDN_<ID> = <SIP Response>,<Q.850 Release Cause> When a SIP response is received (from the IP side), the gateway searches this mapping table for a match. If the SIP response is found, the Release Cause assigned to it is sent to the PSTN. If no match is found, the default static mapping is used. For example: CauseMapSIP2ISDN=404,3 **Note:** This parameter can appear up to 12 times.
CauseMapISDN2SIP_ID [Release Cause Mapping from ISDN to SIP]	Defines a flexible mapping of Q.850 Release Causes and SIP Responses. CauseMapISDN2SIP_<ID> = <Q.850 Release Cause>,<SIP Response> When a Release Cause is received (from the PSTN side), the gateway searches this mapping table for a match. If the Q.850 Release Cause is found, the SIP response assigned to it is sent to the IP side. If no match is found, the default static mapping is used. For example: CauseMapISDN2SIP=6,406 **Note:** This parameter can appear up to 12 times.
RemoveCLIWhenRestricted [Remove CLI when Restricted]	Determines (for IP to Tel calls) whether the Calling Number IE and Calling Name IE are removed from the outgoing ISDN Setup message if the presentation is set to Restricted. 0 = IE aren't removed (default). 1 = IE are removed.

Table 6-9: ISDN and CAS Interworking-Related Parameters (continues on pages 172 to 179)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
ScreeningInd2ISDN [Send Screening Indicator to ISDN]	Overwrites the screening indicator of the calling number for IP→Tel (ISDN) calls. -1 = Not Configured (interworking from IP to ISDN) (default). 0 = User provided, not screened. 1 = User provided, verified and passed. 2 = User provided, verified and failed. 3 = Network provided.
ProgressIndicator2ISDN_ID [Progress Indicator to ISDN]	ID = Trunk number (0-7). 0, 1 or 8 -1 = Not configured (default). If set to '0' PI is not sent to ISDN If set to '1' or '8' the PI value is sent to PSTN in Q.931/Proceeding and Alerting messages. If not configured, the PI in ISDN messages is set according to the 'Play Ringback to Tel' parameter. Usually if PI = 1 or 8, the PSTN/PBX cuts through the audio channel without playing local Ringback tone, enabling the originating party to hear remote Call Progress Tones or network announcements
ProgressIndicator2IP [Progress Indicator to IP]	-1 = (Not configured) for ISDN spans, the progress indicator (PI) that is received in ISDN Proceeding, Progress and Alert messages is used as described in the options below (default). 0 = (No PI) For IP→Tel call, the gateway sends '180 Ringing' SIP response to IP after receiving ISDN Alert or (for CAS) after placing a call to PBX/PSTN. 8, 1 = For IP→Tel call, if 'EnableEarlyMedia=1', the gateway sends '180 Ringing' with SDP in response to an ISDN alert, or it sends a '183 session in progress' message with SDP in response to only the first received ISDN Proceeding or Progress message, after a call is placed to PBX/PSTN over the trunk. This is used to cut through voice path before the remote party answers the call, enabling the originating party to listen to network Call Progress Tones (such as Ringback tone or other network announcements).
PIForDisconnectMsg_ID [Set PI in Rx Disconnect Message]	ID = Trunk number (0-7). Defines the gateway's behavior when a Disconnect message is received from the ISDN before a Connect message was received. -1 (Not configured) = Sends a 183 message according to the received PI in the ISDN Disconnect message. If PI = 1 or 8, the gateway sends a 183 response, enabling the PSTN to play a voice announcement to the IP side. If there isn't a PI in the Disconnect message, the call is released (default). 0 = Do not send a 183 message to IP. The call is released. 1, 8 = Sends 183 message to IP.
ConnectOnProgressInd	0 = Connect message isn't sent after 183 Session Progress is received (default). 1 = Connect message is sent after 183 Session Progress is received. This feature enables the play of announcements from IP to PSTN without the need to answer the Tel→IP call. It can be used with PSTN networks that don't support the opening of a TDM channel before an ISDN Connect message is received.
SIP183Behaviour [183 Message Behavior]	Defines the ISDN message that is sent when 183 Session Progress message is received for IP→Tel calls. 0 = Progress message (default). 1 = Alert message. When set to 1, the gateway sends an Alert message (after the receipt of a 183 response) instead of an ISDN Progress message.
LocalISDNRBSource_ID [Local ISDN Ringback Tone Source]	ID = Trunk number (0-7). Determines whether Ringback tone is played to the ISDN by the PBX / PSTN or by the gateway. 0 = PBX / PSTN (default). 1 = Gateway. This parameter is applicable to ISDN protocols. It is used simultaneously with the parameter 'PlayRBTone2Trunk'.

Table 6-9: ISDN and CAS Interworking-Related Parameters (continues on pages 172 to 179)

ini File Field Name Web Parameter Name	Valid Range and Description
PSTNAlertTimeout [PSTN Alert Timeout]	Alert Timeout in seconds (ISDN T301 timer) for outgoing calls to PSTN. The default is 180 seconds. The range is 0 to 240. **Note:** The PSTN stack T301 timer can be overridden by a lower value, but it can't be increased.
ISDNTransferCapability_ID [ISDN Transfer Capabilities]	ID = Trunk number (0-7). Defines the IP→ISDN Transfer Capability of the Bearer Capability IE in ISDN Setup messages. 0 = Audio 3.1 (default). 1 = Speech. 2 = Data. **Note:** If this parameter isn't configured or equals to '–1', Audio 3.1 capability is used.
SendISDNTransferOnConnect [ISDN Transfer On Connect]	0 = Enable ISDN Transfer if outgoing call is in Alert state (default). 1 = Enable ISDN Transfer only if outgoing call is in Connect state. This parameter is used for the ECT/TBCT/RLT ISDN Transfer methods. Usually, the gateway requests the PBX to connect an incoming and an outgoing call. This parameter determines if the outgoing call (from the gateway to the PBX) must be connected before the transfer is initiated.
ISDNSubAddressFormat	Determines the format of the Subaddress value for ISDN Calling and Called numbers. 0 = ASCII (default). 1 = BCD. For IP-to-Tel calls, if the incoming INVITE message includes 'Subaddress' values for the Called Number (in the Request-URI) and/or the Calling Number (in the From header), these values are interworked to the outgoing ISDN Setup message. If the incoming ISDN SETUP message includes 'subaddress' values for the Called Number and/or the Calling Number, these values are interworked to the outgoing SIP INVITE message.
EnableUUITel2IP [Enable User-to-User IE for Tel to IP]	0 = Disabled (default). 1 = Enable transfer of User-to-User Information Element (UUIE) from PRI Setup message to SIP INVITE message. The IE is transferred using a proprietary 'X-UserToUser' SIP header.
EnableUUIIP2Tel [Enable User-to-User IE for IP to Tel]	0 = Disabled (default). 1 = Enable transfer of (UUIE) from SIP INVITE message to PRI Setup message. The IE is received using a proprietary 'X-UserToUser' SIP header.
ScreeningInd2IP [Send Screening Indicator to IP]	The parameter can overwrite the calling number screening indication for ISDN Tel→IP calls. -1 = not configured (interworking from ISDN to IP) or set to 0 for CAS. 0 = user provided, not screened. 1 = user provided, verified and passed. 2 = user provided, verified and failed. 3 = network provided. **Note:** Applicable only if Remote Party ID (RPID) header is enabled.
SupportRedirectInFacility	0 = Not Supported (default). 1 = Supports partial retrieval of Redirect Number (number only) from a Facility IE in ISDN Setup messages. Applicable to Redirect number according to ECMA-173 Call Diversion Supplementary Services. **Note:** To enable this feature, 'ISDNDuplicateQ931BuffMode' must be set to 1.
EnableCIC	0 = Do not relay the Carrier Identification Code (CIC) to ISDN (default). 1 = CIC is relayed to ISDN in Transit Network Selection IE. If enabled, the CIC code (received in an INVITE Request-URI) is included in a TNS IE in ISDN Setup message. For example: INVITE sip:555666;cic=2345@100.2.3.4 sip/2.0. **Note:** Currently this feature is supported only in SIP→ISDN direction.

Table 6-9: ISDN and CAS Interworking-Related Parameters (continues on pages 172 to 179)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
EnableAOC	0 = Not used (default). 1 = ISDN Advice of Charge (AOC) messages are interworked to SIP. The gateway supports reception of ISDN (Euro ISDN) AOC messages. AOC messages can be received during a call (Facility messages) or at the end of a call (Disconnect or Release messages). The gateway converts the AOC messages into SIP INFO (during a call) and BYE (end of a call) messages using a proprietary AOC SIP header. The gateway supports both Currency and Pulse AOC messages.
TimeForReorderTone	Busy or Reorder Tone duration the CAS gateway plays before releasing the line. The valid range is 0 to 15. The default value is 10 seconds. Applicable also to ISDN if 'PlayBusyTone2ISDN = 2'. Selection of Busy or Reorder tone is done according to release cause received from IP.
DisconnectOnBusyTone [Disconnect Call on Detection of Busy Tone]	0 = Do not disconnect call on detection of busy tone 1 = Disconnect call on detection of busy tone (default). This parameter is applicable to CAS & ISDN protocols.
PlayBusyTone2ISDN [Play Busy Tone to Tel]	This parameter enables the AudioCodes ISDN gateway to play a Busy or a Reorder tone to the PSTN after a call is released. 0 = Immediately sends an ISDN Disconnect message (default). 1 = Sends an ISDN Disconnect message with PI=8 and plays a Busy or a Reorder tone to the PSTN (depending on the release cause). 2 = Delays the sending of an ISDN Disconnect message for 'TimeForReorderTone' seconds and plays a Busy or a Reorder tone to the PSTN. Applicable only if the call is released from the IP before it reaches the Connect state. Otherwise, the Disconnect message is sent immediately and no tones are played.
TrunkTransferMode_X	0 = Not supported (default). 1 = Supports CAS NFA DMS-100 transfer. When a SIP REFER message is received, the gateway performs a Blind Transfer by executing a CAS Wink, waits for an acknowledge Wink from the remote side, dials the Refer-to number to the switch and then releases the call. **Note:** A specific NFA CAS table is required. 2 = Supports ISDN transfer (RLT / TBCT / ECT). When a SIP REFER message is received, the gateway performs a transfer by sending FACILITY messages to the PBX with the necessary information on the call's legs that are to be connected. The different ISDN variants use slightly different methods (using FACILITY messages) to perform the transfer. 3 = Supports CAS Normal transfer. When a SIP REFER message is received, the gateway performs a Blind Transfer by executing a CAS Wink, dialing the Refer-to number to the switch and then releasing the call.
CASTransportType [CAS Transport Type]	0 = Disable CAS relay (default). 1 = Enable CAS relay mode using RFC 2833. The CAS relay mode can be used with the TDM tunneling feature to enable tunneling over IP for both voice and CAS signaling bearers.
CASAddressingDelimiters	Determines if delimiters are added to the dialed address or dialed ANI parameters. Valid options include: 0 = Disable (default) 1 = Enable When this parameter is enabled, delimiters such as '*', '#', and 'ST' are added to the dialed address or dialed ANI parameters. When it is disabled, the address and ANI strings remain without delimiters.

Table 6-9: ISDN and CAS Interworking-Related Parameters (continues on pages 172 to 179)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
EnableVoiceDetection	1 = The gateway sends 200 OK message (in response to INVITE) when speech/fax/modem is detected. 0 = The gateway sends 200 OK message (in response to INVITE) immediately after the gateway finishes dialing (default). **Note 1:** To activate this feature set the parameter 'EnableDSPIPMDetectors' to 1. Usually this feature is used only when early media is used to establish voice path before the call is answered. **Note 2:** This feature is applicable only when the protocol type is CAS.
EnableDSPIPMDetectors	Enables / disables the IPmedia DSP detectors. 0 = Disable (default). 1 = Enable. **Note1:** The gateway's Feature Key should contain the "IPMDetector" DSP option. **Note 2:** When EnableDSPIPMDetectors = 1, the number of available gateway channels is reduced by a factor of 5/6. For example, a gateway with 8 E1 spans, capacity is reduced to 6 spans (180 channels), while a gateway with 8 T1 spans, capacity remains the same (192 channels).
XChannelHeader	0 = x-channel header is not used (default). 1 = x-channel header is generated, with trunk/B-channel information. The header provides information on the E1/T1 physical trunk/B-channel on which the call is received or placed. For example 'x-channel: DS/DS1-5/22'. This header is generated by the gateway and is sent in the following messages: INVITE and 183/180/200OK responses.
AddIEinSetup [Add IE in SETUP]	This parameter enables to add an optional Information Element data (in hex format) to ISDN SETUP message. For example: to add the following IE: '0x20,0x02,0x00,0xe1', define: 'AddIEinSetup = 200200e1'. **Note:** This IE is sent from the Trunk Group IDs defined by the parameter 'SendIEonTG'.
SendIEonTG [Trunk Groups to Send IE]	A list of Trunk Group IDs (up to 50 characters) from where the optional ISDN IE, defined by the parameter 'AddIEinSetup', is sent. For example: 'SendIEonTG = 1,2,4,10,12,6'.
ISDNDMSTimerT310	Overrides the T310 timer for the DMS-100 ISDN variant. T310 defines the timeout between the reception of Proceeding message and the reception of Alert / Connect message. The valid range is 10 to 30. The default value is 10 (seconds). **Note:** Applicable only to Nortel DMS and Nortel MERIDIAN PRI variants (ProtocolType = 14 and 35).
ISDNJapanNTTTimerT3JA	T3_JA timer (in seconds). This parameter overrides the internal PSTN T301 timeout on the Users Side (TE side). If an outgoing call from the gateway to ISDN is not answered during this timeout, the call is released. The valid range is 10 to 240. The default value is 50. Applicable only to Japan NTT PRI variant (ProtocolType = 16). **Note:** This timer is also affected by the parameter 'PSTNAlertTimeout'.
EarlyAnswerTimeout	Defines the time (in seconds) the gateway waits for a CONNECT response from the called party (Tel side) after sending a SETUP message. If the timer expires, the call is answered by sending a 200 OK message (IP side). The valid range is 0 to 600. The default value is 0 (disable).

Table 6-9: ISDN and CAS Interworking-Related Parameters (continues on pages 172 to 179)

ini File Field Name Web Parameter Name	Valid Range and Description
EnablePatternDetector [Enable Pattern Detector]	Enables or disables activation of the Pattern Detector (PD). Valid options include: 0 = Disable (default). 1 = Enable.
PDThreshold	Defines the number of consecutive patterns to trigger the pattern detection event. The valid range is 0 to 31. The default is 5.
PDPattern	Defines the patterns that can be detected by the Pattern Detector. The valid range is 0 to 0xFF.

6.15 Number Manipulation and Routing Parameters

Table 6-10: Number Manipulation and Routing Parameters (continues on pages 180 to 188)

ini File Field Name Web Parameter Name	Valid Range and Description
TrunkGroup_x [Trunk Group Table]	TrunkGroup_x = T-U/a-b,c,d x = Trunk group ID (1 to 99). T = Starting physical trunk number (0 to 7). U = Ending physical trunk number (0 to 7). a = Starting B-channel (from 1). b = Ending B-channel (up to 31). **Note:** To represent all B-channels use a single asterisk instead of 'a' and 'b'. c = Phone number associated with the first channel (optional). d = Optional Tel Profile ID (1 to 4). For example: TrunkGroup_1 = 0/1-31,1000 (for E1 span). TrunkGroup_1 = 1/1-31,$$,1. TrunkGroup_2 = 2/1-24,3000 (for T1 span). TrunkGroup_1 = 0-7/1-20,1000 (for 8 E1 spans, 20 B-channels). TrunkGroup_1 = 0-3/*,1000 (for 4 E1 spans, all B-channels). Trunk group is the recommended method to configure the gateway's B-channels. **Note:** An optional Tel Profile ID (1 to 9) can be applied to each group of B-channels.
ChannelList	This parameter is obsolete; use instead TrunkGroup_x.
DefaultNumber [Default Destination Number]	Defines the telephone number that the gateway uses if the parameter 'TrunkGroup_x' doesn't include a phone number. The parameter is used as a starting number for the list of B-channels comprising all trunk groups in the gateway.

Table 6-10: Number Manipulation and Routing Parameters (continues on pages 180 to 188)

ini File Field Name Web Parameter Name	Valid Range and Description
ChannelSelectMode [Channel Select Mode]	Defines common rule of port allocation for IP to TEL calls. • 0 = By phone number - Select the gateway port according to the called number (refer to the note below). • 1 = Cyclic Ascending - Select the next available channel in an ascending cycle order. Always select the next higher channel number in the Trunk Group. When the gateway reaches the highest channel number in the Trunk Group, it selects the lowest channel number in the Trunk Group and then starts ascending again (default). • 2 = Ascending - Select the lowest available channel. Always start at the lowest channel number in the Trunk Group and if that channel is not available, select the next higher channel. • 3 = Cyclic Descending - Select the next available channel in descending cycle order. Always select the next lower channel number in the Trunk Group. When the gateway reaches the lowest channel number in the Trunk Group, it selects the highest channel number in the Trunk Group and then start descending again. • 4 = Descending - Select the highest available channel. Always start at the highest channel number in the Trunk Group and if that channel is not available, select the next lower channel. • 5 = Number + Cyclic Ascending – First select the gateway port according to the called number (refer to the note below). If the called number isn't found, then select the next available channel in ascending cyclic order. Note that if the called number is found, but the port associated with this number is busy, the call is released. • 6 = By Source Phone Number - Select the gateway port according to the calling number (refer to the note below). **Note:** The internal numbers of the gateway's B-channels are defined by the 'TrunkGroup_x' parameter (under 'Phone Number').
TrunkGroupSettings [Trunk Group Settings]	Defines rules for port allocation for specific Trunk Groups. If no rule exists, the global rule defined by ChannelSelectMode applies. The format is: a, b, c Where, a = Trunk Group ID number. b = Channel select mode for that Trunk Group. c = Registration mode for that Trunk Group (Per Endpoint [0] or Per Trunk Group [1]). If not configured [-1], the value of AuthenticationMode is used. Available values are identical to those defined by the ChannelSelectMode parameter.
AddTrunkGroupAsPrefix [Add Trunk Group ID as Prefix]	0 = not used 1 = For Tel→IP incoming call, Trunk Group ID is added as prefix to destination phone number. Applicable only if trunk group ID are configured. Can be used to define various routing rules.
AddPortAsPrefix [Add Trunk ID as Prefix]	0 = Don't add (default) 1 = Add trunk ID number (single digit in the range 1 to 8) as a prefix to the called phone number for Tel→IP incoming calls. This option can be used to define various routing rules.
ReplaceEmptyDstWithPort Number [Replace Empty Destination with Port Number]	0 = Disabled (default). 1 = Enabled, Internal channel number is used as a destination number if called number is missing. **Note:** Applicable only to Tel→IP calls, if called number is missing.
CopyDestOnEmptySource	0 = Leave Source Number empty (default). 1 = If the Source Number of an incoming Tel to IP call is empty, the Destination Number is copied to the Source Number.

Table 6-10: Number Manipulation and Routing Parameters (continues on pages 180 to 188)

ini File Field Name Web Parameter Name	Valid Range and Description
AddNPIandTON2CallingNumber [Add NPI and TON to Calling Number]	0 = Do not change the Calling Number (default). 1 = Add NPI and TON to the Calling Number of incoming (Tel to IP) ISDN call. For example: After receiving a Calling Number = 555, NPI = 1 and TON = 3, the modified number is going to be 13555. This number can later be used for manipulation and routing purposes.
AddNPIandTON2CalledNumber [Add NPI and TON to Called Number]	0 = Do not change the Called Number (default). 1 = Add NPI and TON to the Called Number of incoming (Tel to IP) ISDN call. For example: After receiving a Called Number = 555, NPI=1 and TON = 3, the modified number is going to be 13555. This number can later be used for manipulation and routing purposes.
UseSourceNumberAsDisplayName [Use Source Number as Display Name]	Applicable to Tel→IP calls. 0 = The Tel Source Number is used as the IP Source Number and the Tel Display Name is used as the IP Display Name (if Tel Display Name is received). If no Display Name is received from the Tel side, the IP Display Name remains empty (default). 1 = If a Tel Display Name is received, the Tel Source Number is used as the IP Source Number and the Tel Display Name is used as the IP Display Name. If no Display Name is received from the Tel side, the Tel Source Number is used as the IP Source Number and also as the IP Display Name. 2 = The Tel Source Number is used as the IP Source Number and also as the IP Display Name (even if the received Tel Display Name is not empty).
UseDisplayNameAsSourceNumber [Use Display Name as Source Number]	Applicable to IP→Tel calls. 0 = The IP Source Number is used as the Tel Source Number and the IP Display Name is used as the Tel Display Name (if IP Display Name is received). If no Display Name is received from IP, the Tel Display Name remains empty (default). 1 = If an IP Display Name is received, it is used as the Tel Source Number and also as the Tel Display Name, the Presentation is set to Allowed (0). If no Display Name is received from IP, the IP Source Number is used as the Tel Source Number and the Presentation is set to Restricted (1). For example: When the following is received 'from: 100 <sip:200@201.202.203.204>', the outgoing Source Number and Display Name are set to '100' and the Presentation is set to Allowed (0). When the following is received 'from: <sip:100@101.102.103.104>', the outgoing Source Number is set to '100' and the Presentation is set to Restricted (1).
AlwaysUseRouteTable [Use Routing Table for Host Names and Profiles]	Use the internal Tel to IP routing table to obtain the URI Host name and (optionally) an IP profile (per call), even if Proxy server is used. 0 = Don't use (default) 1 = Use **Note:** This Domain name is used, instead of Proxy name or Proxy IP address, in the INVITE SIP URI.

Table 6-10: Number Manipulation and Routing Parameters (continues on pages 180 to 188)

ini File Field Name Web Parameter Name	Valid Range and Description
Prefix [Tel to IP Routing Table]	Prefix = <Destination Phone Prefix>,<Destination IP Address>,<Source Phone Prefix>,<Profile ID> For example: Prefix = 20,10.2.10.2,202,1 Prefix = 10[340-451]xxx#,10.2.10.6,*,1 Prefix = *,gateway.domain.com,* **Note 1:** <destination / source phone prefix> can be single number or a range of numbers. **Note 2:** This parameter can appear up to 50 times. **Note 3:** Parameters can be skipped by using the sign '$$', for example: Prefix = $$,10.2.10.2,202,1 **Note 4:** The <Destination IP Address> field can be either in dotted format notation or a FQDN. This field can also include a selected port to use (in the format: <IP Address>:<Port>). **Note 5:** The IP address can include wildcards. The 'x' wildcard is used to represent single digits, e.g., 10.8.8.xx represents all addresses between 10.8.8.10 to 10.8.8.99. The '*' wildcard represents any number between 0 and 255, e.g., 10.8.8.* represents all addresses between 10.8.8.0 and 10.8.8.255. **Note 6:** If the string 'ENUM' is specified in the Destination IP Address field, an ENUM query containing the destination phone number is sent to the DNS server. The ENUM reply includes a SIP URI, used as the Request-URI in the outgoing INVITE and for routing (if Proxy is not used). For available notations, refer to Section 5.5.3.1 on page 67. For detailed information on this feature, refer to Section 5.5.5.1 on page 70.
PSTNPrefix [IP to Trunk Group Routing Table]	PSTNPrefix = a,b,c,d,e a = Destination Number Prefix b = Trunk group ID (1 to 99) c = Source Number Prefix d = Source IP address (obtained from the Contact header in the INVITE message) e = IP Profile ID (1 to 4) Selection of trunk groups (for IP to Tel calls) is according to destination number, source number and source IP address. **Note 1:** To support the 'in call alternative routing' feature, users can use two entries that support the same call, but assigned it with a different trunk groups. The second entree functions as an alternative selection if the first rule fails as a result of one of the release reasons listed in the AltRouteCauseIP2Tel table. **Note 2:** An optional IP ProfileID (1 to 4) can be applied to each routing rule. **Note 3:** The Source IP Address can include the 'x' wildcard to represent single digits. For example: 10.8.8.xx represents all IP addresses between 10.8.8.10 to 10.8.8.99. The '*' wildcard represents any number between 0 and 255, e.g., 10.8.8.* represents all addresses between 10.8.8.0 and 10.8.8.255. **Note 4:** If the Source IP field includes an FQDN, DNS resolution is performed according to DNSQueryType.

Table 6-10: Number Manipulation and Routing Parameters (continues on pages 180 to 188)

ini File Field Name Web Parameter Name	Valid Range and Description
RemovePrefix [IP to Tel Remove Routing Table Prefix]	0 = Don't remove prefix (default) 1 = Remove PSTN prefix (defined in the IP to Trunk Group routing table) from a telephone number of an incoming IPàTel call, before forwarding it to PSTN. For example: To route an incoming IPàTel Call with destination number 21100, the IP to Trunk Group Routing table is scanned for a matching prefix. If such prefix is found, 21 for instance, then before the call is routed to the corresponding trunk group the prefix (21) is removed from the original number, so that only 100 is left. Note: Applicable only if number manipulation is performed after call routing for IPàTel calls (RouteModeIP2Tel = 0).
RouteModeIP2Tel [IP to Tel routing Mode]	0 = Route calls before number manipulation (default) 1 = Route calls after number manipulation Defines order between routing calls to Trunk group and manipulation of destination number
RouteModeTel2IP [Tel to IP routing Mode]	0 = Route calls before number manipulation (default) 1 = Route calls after number manipulation Defines order between routing incoming calls to IP, using routing table, and manipulation of destination number Not applicable if Outbound Proxy is used.
SwapRedirectNumber [Swap Redirect and Called Numbers]	0 = Don't change numbers (default) 1 = Incoming ISDN call that includes redirect number (sometimes referred as 'original called number') uses this number instead of the called number.
Prefix2RedirectNumber [Add Prefix to Redirect Number]	Defines a string Prefix that is added to the Redirect number received from the Tel side. This Prefix is added to the Redirect Number in the Diversion header. The valid range is an 8-character string. The default is an empty string.
AddTON2RPI [Add Number Plan and Type to Remote Party ID Header]	0 = TON/PLAN parameters aren't included in the RPID header. 1 = TON/PLAN parameters are included in the RPID header (default). If RPID header is enabled (EnableRPIHeader = 1) and 'AddTON2RPI=1', it is possible to configure the calling and called number type and number plan using the Number Manipulation tables for Tel→IP calls.
NumberMapTel2IP [Destination Phone Number Manipulation Table for Tel→IP calls]	Manipulates the destination number for Tel to IP calls. NumberMapTel2IP = a,b,c,d,e,f,g a = Destination number prefix b = Number of stripped digits from the left, or (if brackets are used) from the right. A combination of both options is allowed. c = String to add as prefix, or (if brackets are used) as suffix. A combination of both options is allowed. d = Number of remaining digits from the right e = Number Plan used in RPID header f = Number Type used in RPID header g = Source number prefix The 'b' to 'f' manipulations rules are applied if the called and calling numbers match the 'a' and 'g' conditions. The manipulation rules are executed in the following order: 'b', 'd' and 'c'. Parameters can be skipped by using the sign '$$', for example: NumberMapTel2IP=01,2,972,$$,0,0,$$ NumberMapTel2IP=03,(2),667,$$,0,0,22

Table 6-10: Number Manipulation and Routing Parameters (continues on pages 180 to 188)

ini File Field Name Web Parameter Name	Valid Range and Description
NumberMapIP2Tel [Destination Phone Number Manipulation Table for IP→Tel calls]	Manipulate the destination number for IP to Tel calls. NumberMapIP2Tel = a,b,c,d,e,f,g,h,i a = Destination number prefix b = Number of stripped digits from the left, or (if brackets are used) from the right. A combination of both options is allowed. c = String to add as prefix, or (if brackets are used) as suffix. A combination of both options is allowed. d = Number of remaining digits from the right e = Q.931 Number Plan f = Q.931 Number Type g = Source number prefix h = Not applicable, set to $$ i = Source IP address (obtained from the Contact header in the INVITE message) The 'b' to 'f' manipulation rules are applied if the called and calling numbers match the 'a', 'g' and 'i' conditions. The manipulation rules are executed in the following order: 'b', 'd' and 'c'. Parameters can be skipped by using the sign '$$', for example: NumberMapIP2Tel =01,2,972,$$,0,$$,034 NumberMapIP2Tel =03,(2),667,$$,$$,0,22,$$,10.13.77.8 **Note:** The Source IP address can include the 'x' wildcard to represent <u>single</u> digits. For example: 10.8.8.xx represents all the addresses between 10.8.8.10 to 10.8.8.99. The '*' wildcard represents any number between 0 and 255, e.g. 10.8.8.* represents all the addresses between 10.8.8.0 and 10.8.8.255.
SourceNumberMapTel2IP [Source Phone Number Manipulation Table for Tel→IP calls]	SourceNumberMapTel2IP = a,b,c,d,e,f,g,h a = Source number prefix b = Number of stripped digits from the left, or (if in brackets are used) from right. A combination of both options is allowed. c = String to add as prefix, or (if in brackets are used) as suffix. A combination of both options is allowed. d = Number of remaining digits from the right e = Number Plan used in RPID header f = Number Type used in RPID header g =Destination number prefix h =Calling number presentation (0 to allow presentation, 1 to restrict presentation) The 'b' to 'f' and 'h' manipulation rules are applied if the called and calling numbers match the 'a' and 'g' conditions. The manipulation rules are executed in the following order: 'b', 'd' and 'c'. Parameters can be skipped by using the sign '$$', for example: SourceNumberMapTel2IP=01,2,972,$$,0,0,$$,1 SourceNumberMapTel2IP=03,(2),667,$$,0,0,22,0

Table 6-10: Number Manipulation and Routing Parameters (continues on pages 180 to 188)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
SourceNumberMapIP2Tel [Source Phone Number Manipulation Table for IP→Tel calls]	Manipulate the source number for IP to Tel calls. SourceNumberMapIP2Tel = a,b,c,d,e,f,g,h,i a = Source number prefix b = Number of stripped digits from the left, or (if brackets are used) from the right. A combination of both options is allowed. c = String to add as prefix, or (if brackets are used) as suffix. A combination of both options is allowed. d = Number of remaining digits from the right e = Q.931 Number Plan f = Q.931 Number Type g = Destination number prefix h =Calling number presentation (0 to allow presentation, 1 to restrict presentation) I = Source IP address (obtained from the Request-URI in the INVITE message). The 'b' to 'f' and 'h' manipulation rules are applied if the called and calling numbers match the 'a', 'g' and 'I' conditions. The manipulation rules are executed in the following order: 'b', 'd' and 'c'. Parameters can be skipped by using the sign '$$', for example: SourceNumberMapIP2Tel =01,2,972,$$,0,$$,034,1 SourceNumberMapIP2Tel =03,(2),667,$$,$$,0,22 **Note:** The Source IP address can include the 'x' wildcard to represent single digits. For example: 10.8.8.xx represents all the addresses between 10.8.8.10 to 10.8.8.99. The '*' wildcard represents any number between 0 and 255, e.g. 10.8.8.* represents all the addresses between 10.8.8.0 and 10.8.8.255.
For ETSI ISDN variant, the following Number Plan and Type combinations (Plan/Type) are supported in the Destination and Source Manipulation tables: 0,0 = Unknown, Unknown 9,0 = Private, Unknown 9,1 = Private, Level 2 Regional 9,2 = Private, Level 1 Regional 9,3 = Private, PISN Specific 9,4 = Private, Level 0 Regional (local) 1,0 = Public(ISDN/E.164), Unknown 1,1 = Public(ISDN/E.164), International 1,2 = Public(ISDN/E.164), National 1,3 = Public(ISDN/E.164), Network Specific 1,4 = Public(ISDN/E.164), Subscriber 1,6 = Public(ISDN/E.164), Abbreviated For NI-2 and DMS-100 ISDN variants the valid combinations of TON and NPI for calling and called numbers are (Plan/Type): 0/0 - Unknown/Unknown 1/1 - International number in ISDN/Telephony numbering plan 1/2 - National number in ISDN/Telephony numbering plan 1/4 - Subscriber (local) number in ISDN/Telephony numbering plan 9/4 - Subscriber (local) number in Private numbering plan	
SecureCallsFromIP [IP Security]	0 = Gateway accepts all SIP calls (default). 1 = Gateway accepts SIP calls only from IP addresses defined in the Tel to IP routing table. The gateway rejects all calls from unknown IP addresses. For detailed information on the Tel to IP Routing table, refer to Section 5.5.5.1 on page 70. **Note:** Specifying the IP address of a Proxy server in the Tel to IP Routing table enables the gateway to only accept calls originating in the Proxy server and rejects all other calls.

Table 6-10: Number Manipulation and Routing Parameters (continues on pages 180 to 188)

ini File Field Name Web Parameter Name	Valid Range and Description
AltRouteCauseTel2IP [Reasons for Alternative Routing Table]	Table of call failure reason values received from the IP side. If a call is released as a result of one of these reasons, the gateway tries to find an alternative route to that call in the 'Tel to IP Routing' table. For example: AltRouteCauseTel2IP = 486 (Busy here). AltRouteCauseTel2IP = 480 (Temporarily unavailable). AltRouteCauseTel2IP = 408 (No response). **Note 1:** The 408 reason can be used to specify that there was no response from the remote party to the INVITE request. **Note 2:** This parameter can appear up to 5 times.
AltRouteCauseIP2Tel [Reasons for Alternative Routing Table]	Table of call failure reason values received from the PSTN side (in Q.931 presentation). If a call is released as a result of one of these reasons, the gateway tries to find an alternative trunk group to that call in the 'IP to Trunk Group Routing' table. For example: AltRouteCauseIP2Tel = 3 (No route to destination). AltRouteCauseIP2Tel = 1 (Unallocated number). AltRouteCauseIP2Tel = 17 (Busy here). **Note 1:** This parameter can appear up to 5 times. **Note 2**: If the gateway fails to establish a cal to the PSTN because it has no available channels in a specific trunk group (e.g., all of the trunk group's channels are occupied, or the trunk group's spans are disconnected or out of sync), it uses the internal release cause '3' (no route to destination). This cause can be used in the 'AltRouteCauseIP2Tel' table to define routing to an alternative trunk group.
FilterCalls2IP [Filter Calls To IP]	0 = Disabled (default) 1 = Enabled If the filter calls to IP feature is enabled, then when a Proxy is used, the gateway first checks the Tel→IP routing table before making a call through the Proxy. If the number is not allowed (number isn't listed or a Call Restriction routing rule, IP=0.0.0.0, is applied), the call is released.
Alternative Routing Parameters	
AltRoutingTel2IPEnable [Enable Alt Routing Tel to IP]	Operation modes of the Alternative Routing mechanism: 0 = Disabled (default). 1 = Enabled. 2 = Enabled for status only, not for routing decisions.
AltRoutingTel2IPMode [Alt Routing Tel to IP Mode]	0 (None) = Alternative routing is not used. 1 (Conn) = Alternative routing is performed if ping to initial destination failed. 2 (QoS) = Alternative routing is performed if poor quality of service was detected. 3 (All) = Alternative routing is performed if, either ping to initial destination failed, or poor quality of service was detected, or DNS host name is not resolved (default). **Note:** QoS is quantified according to delay and packet loss, calculated according to previous calls. QoS statistics are reset if no new data is received for two minutes. For information on the Alternative Routing feature, refer to Section 8.3 on page 210.
IPConnQoSMaxAllowedPL [Max Allowed Packet Loss for Alt Routing]	Packet loss percentage at which the IP connection is considered a failure. The range is 1% to 20%. The default value is 20%.

Table 6-10: Number Manipulation and Routing Parameters (continues on pages 180 to 188)

ini File Field Name Web Parameter Name	Valid Range and Description
IPConnQoSMaxAllowedDelay [Max Allowed Delay for Alt Routing]	Transmission delay (in msec) at which the IP connection is considered a failure. The range is 100 to 1000. The default value is 250 msec.
Alternative Routing Parameters	
AddPhoneContextAsPrefix [Add Phone Context As Prefix]	Determines whether or not the received Phone-Context parameter is added as a prefix to the outgoing ISDN SETUP Called and Calling numbers. Valid options include: 0 = Disable (default). 1 = Enable.
PhoneContext [Phone Context Table]	When a call is received from the ISDN, the NPI and TON are compared against the table, and the Phone-Context value is used in the outgoing SIP INVITE message. The same mapping occurs when an INVITE with a Phone-Context attribute is received. The Phone-Context parameter appears in the standard SIP headers where a phone number is used (Request-URI, To, From, Diversion). PhoneContext = <Number Plan>,<Number Type>,<Phone-Context> For example: ▪ PhoneContext = 0,0,unknown.com ▪ PhoneContext = 1,1,host.com ▪ PhoneContext = 9,1,na.e164.host.com **Note 1:** This parameter can appear up to 20 times. **Note 2:** Several rows with the same NPI-TON or Phone-Context are allowed. In this scenario, a Tel-to-IP call uses the first match. **Note 3:** Phone-Context '+' is a unique case as it doesn't appear in the Request-URI as a Phone-Context parameter. Instead, it's added as a prefix to the phone number. The '+' isn't removed from the phone number in the IP-to-Tel direction.

6.16 E1/T1 Configuration Parameters

Table 6-11: E1/T1/J1 Configuration Parameters (continues on pages 189 to 195)

ini File Field Name Web Parameter Name	Valid Range and Description
TDMBusType	TP-260 only. Must be set to 2.
PCMLawSelect [PCM Law Select]	1 = A-law (default) 3 = μ-Law Usually A-Law is used for E1 spans and μ-Law for T1 and J1 spans.
ProtocolType [Protocol Type]	Sets the PSTN protocol to be used for this trunk. E1_EURO_ISDN = 1 T1_CAS = 2 T1_RAW_CAS = 3 T1_TRANSPARENT = 4 E1_TRANSPARENT_31 = 5 E1_TRANSPARENT_30 = 6 E1_MFCR2 = 7 E1_CAS_R2 = 8 E1_RAW_CAS = 9 T1_NI2_ISDN = 10 T1_4ESS_ISDN = 11 T1_5ESS_9_ISDN = 12 T1_5ESS_10_ISDN = 13 T1_DMS100_ISDN = 14 J1_TRANSPARENT = 15 T1_NTT_ISDN = 16 (Japan - Nippon Telegraph) E1_AUSTEL_ISDN = 17 (Australian Telecom) T1_HKT_ISDN = 18 (Hong Kong – HKT) E1_KOR_ISDN = 19 (Korean operator) T1_HKT_ISDN = 20 (Hong Kong - HKT over T1) E1_QSIG = 21 T1_QSIG = 23 E1_FRENCH_VN3_ISDN = 31 T1_DMS100_Meridian = 35 E1_NI2_ISDN = 40 **Note:** The gateway simultaneously supports different variants of CAS and PRI protocols on different E1/T1 spans (no more than four simultaneous PRI variants).
ProtocolType_x [Protocol Type]	Same as the description for parameter 'ProtocolType' for a specific trunk ID (x = 0 to 7).
TraceLevel [Trace Level]	Defines the trace level. Valid options include: 0 = No trace (default) 1= Full ISDN trace 2 = Layer 3 ISDN trace 3 = Only ISDN Q.931 messages trace 4 = Layer 3 ISDN no duplication trace

Table 6-11: E1/T1/J1 Configuration Parameters (continues on pages 189 to 195)

ini File Field Name Web Parameter Name	Valid Range and Description
FramingMethod [Framing Method]	Selects the framing method to be used for E1/T1 spans. **For E1** 0 = Multiframe with CRC4 (default, automatic mode, if CRC is identified in the Rx, CRC is sent in Tx, otherwise no CRC). a = Double frame c = Multiframe with CRC4 **For T1** 0 or D = Extended super frame with CRC6 (default) 1 or B = Super frame D4, F12 (12-Frame multiframe) A = F4 (4-Frame multiframe) C = Extended super frame without CRC6 F = J1 - Japan (ESF with CRC6 and JT)
FramingMethod_x [Framing Method]	Same as the description for parameter 'FramingMethod' for a specific trunk ID (x = 0 to 7).
TerminationSide [ISDN Termination Side]	Selects the ISDN termination side. Applicable only to ISDN protocols. 0 = ISDN User Termination Side (TE) (default) 1 = ISDN Network Termination Side (NT) **Note:** select 'User Side' when the PSTN or PBX side is configured as 'Network side', and vice-versa. If you don't know the gateway's ISDN termination side, choose 'User Side' and refer to the 'Status & Diagnostics>Channel Status' screen. If the D-channel alarm is indicated, choose 'Network Side'.
TerminationSide_x [ISDN Termination Side]	Same as the description for parameter 'TerminationSide' for a specific trunk ID (x = 0 to 7).
ClockMaster [Clock Master]	Determines the Tx clock source of the E1/T1 line. 0 = Generate the clock according to the Rx of the E1/T1 line (default). 1 = Generate the clock according to the internal TDM bus. For detailed information on configuring the gateway's clock settings, refer to Section 10.1 on page 243. **Note:** The source of the internal TDM bus clock is determined by the parameter TDMBusClockSource.
ClockMaster_x [Clock Master]	Same as the description for parameter 'ClockMaster' for a specific trunk ID (x = 0 to 7).
TDMBusClockSource [TDM Bus Clock Source]	1 = Generate clock from local source (default). 4 = Recover clock from PSTN line. For detailed information on configuring the gateway's clock settings, refer to Section 10.1 on page 243.
TDMBusPSTNAutoClockEnable [TDM Bus PSTN Auto Clock]	0 = Recovers the clock from the E1/T1 line defined by the parameter TDMBusLocalReference (default) 1 = Recovers the clock from any connected synchronized slave E1/T1 line. If this trunk loses its synchronization, the gateway attempts to recover the clock from the next trunk. Note that initially the gateway attempts to recover the clock from the trunk defined by the parameter TDMBusLocalReference. **Note:** This parameter is relevant only if 'TDMBusClockSource = 4'
TDMBusLocalReference [TDM Bus Local Reference]	0 to 7 (default = 0) Physical Trunk ID from which the gateway recovers its clock. Applicable only if 'TDMBusClockSource = 4' and 'TDMBusPSTNAutoClockEnable = 0'
AutoClockTrunkPriority [Auto Clock Trunk Priority]	Defines the trunk priority for auto-clock fallback (per trunk parameter). 0 to 99 = priority (0 is the highest = default). 100 = the SW never performs a fallback to that trunk (usually used to mark un-trusted source of clock). **Note:** Fallback is enabled when the TDMBusPSTNAutoClockEnable parameter is set to 1.

Table 6-11: E1/T1/J1 Configuration Parameters (continues on pages 189 to 195)

ini File Field Name Web Parameter Name	Valid Range and Description
TDMBusPSTNAutoClockRevertingEnable [TDM Bus PSTN Auto Clock Reverting]	Enables and disables the PSTN trunk auto-fallback reverting feature. If a trunk with a higher priority than the current LocalReference is being synchronized, the board LocalReference changes to the new trunk. 0 = disable (default). 1 = enable. **Note:** The parameter is valid only in when the TDMBusPSTNAutoClockEnable parameter is set to 1.
LineCode [Line Code]	0 = use B8ZS line code (for T1 trunks only) default. 1 = use AMI line code. 2 = use HDB3 line code (for E1 trunks only). Use to select B8ZS or AMI for T1 spans, and HDB3 or AMI for E1 spans.
LineCode_x [Line Code]	Same as the description for parameter 'LineCode' for a specific trunk ID (x = 0 to 7).
BchannelNegotiation [B-channel Negotiation]	Determines the ISDN B-Channel negotiation mode. 0 = Preferred 1 = Exclusive (default) Applicable to ISDN protocols.
NFASGroupNumber_x [NFAS Group Number]	0 = Non NFAS trunk (default) 1 to 4 = NFAS group number Indicates the NFAS group number (NFAS member) for the selected trunk. 'x' identifies the Trunk ID (0-7). Trunks that belong to the same NFAS group have the same number. With ISDN Non-Facility Associated Signaling you can use single D-channel to control multiple PRI interfaces. Applicable only to T1 ISDN protocols.
DchConfig_x [D-channel Configuration]	0 = Primary Trunk (default) 1 = Backup Trunk 2 = NFAS Trunk D-channel configuration parameter defines primary, backup (optional) and B-channels only trunks. 'x' identifies the Trunk ID (0-7). Primary trunk contains D-channel that is used for signaling. Backup trunk contains backup D-channel that is used if the primary D-channel fails. The other NFAS trunks contain only 24 B-channels, without a signaling D-channel. **Note:** Applicable only to T1 ISDN protocols.
ISDNNFASInterfaceID_x [NFAS Interface ID]	Defines a different Interface ID for each T1 trunk. The valid range is 0 to 100. The default interface ID equals to the trunk's ID (0 to 7). 'x' identifies the trunk ID (0-7) **Note:** To set the NFAS interface ID, configure: ISDNIBehavior_x to include '512' feature, per T1 trunk.
CASTableIndex_x [CAS Table]	Defines CAS protocol for each trunk ID (x = 0 to 7) from a list of CAS protocols defined by the parameter CASFileName_Y. For example: CASFileName_0 = 'E_M_WinkTable.dat' CASFileName_1 = 'E_M_ImmediateTable.dat' CASTableIndex_0 = 0 CASTableIndex_1 = 0 CASTableIndex_2 = 1 CASTableIndex_3 = 1 Trunks 0 and 1 use the E&M Winkstart CAS protocol, while trunks 2 and 3 use the E&M Immediate Start CAS protocol.

Table 6-11: E1/T1/J1 Configuration Parameters (continues on pages 189 to 195)

ini File Field Name Web Parameter Name	Valid Range and Description
CASFileName_0 **CASFileName_1** **CASFileName_7**	CAS file name (e.g., 'E_M_WinkTable.dat') that defines the CAS protocol. It is possible to define up to eight different CAS files by repeating this parameter. Each CAS file can be associated with one or more of the gateway trunks using the parameter CASTableIndex_x.
CASTablesNum	1 to 8. Indicates how many CAS protocol configurations files are loaded.
IdleABCDPattern [Idle ABCD Pattern]	Range 0x0 to 0xF Default = -1 (default pattern = 0000) ABCD (CAS) Pattern to be applied to CAS signaling bus when the channel is idle. This is only relevant when using PSTN interface with CAS protocols. Set to -1 for default.
IdlePCMPattern [Idle PCM Pattern]	Range 0x00 to 0xFF Default = -1 (default pattern = 0xFF for μ-Law, 0x55 for A-law) PCM Pattern to be applied to E1/T1 timeslot (B-channel) when the channel is idle.
LineBuildOut.Loss [Line Build Out Loss]	0 = 0 dB (default) 1 = -7.5 dB 2 = -15 dB 3 = -22.5 dB Selects the line build out loss to be used for T1 trunks N/A for E1 trunks.
ISDNRxOverlap_x [Enable Receiving of Overlap Dialing]	Enable / disable Rx ISDN overlap per trunk ID (x = 0 to 7). 0 = Disabled (default). 1 = Enabled. **Note 1:** If enabled, the gateway receives ISDN called number that is sent in the 'Overlap' mode. **Note 2:** The SETUP message to IP is sent only after the number (including the 'Sending Complete' Info Element) was fully received (via SETUP and/or subsequent INFO Q.931 messages). **Note3:** The 'MaxDigits' parameter can be used to limit the length of the collected number for gateway ISDN overlap dialing (if sending complete was not received). **Note 4:** If a digit map pattern is defined (DigitMapping), the gateway collects digits until a match is found (e.g., useful for closed numbering schemes) or until a timer expires (e.g., useful for open numbering schemes). If a match is found (or the timer expires), the digit collection process is terminated even if Sending Complete wasn't received.
ISDNRxOverlap	0 = Disabled (default). 1 = Enabled. Any number bigger than one = Number of digits to receive. **Note 1:** If enabled, the gateway receives ISDN called number that is sent in the 'Overlap' mode. **Note 2:** The INVITE to IP is sent only after the number (including 'Sending Complete' Info Element) was fully received (in SETUP and/or subsequent INFO Q.931 messages). For detailed information on ISDN overlap dialing, refer to Section 10.1 on page 243.

Table 6-11: E1/T1/J1 Configuration Parameters (continues on pages 189 to 195)

ini File Field Name Web Parameter Name	Valid Range and Description
DigitMapping [Digit Mapping Rules]	Digit map pattern (used to reduce the dialing period when Overlap dialing is used). If the digit string (dialed number) has matched one of the patterns in the digit map, the gateway stops collecting digits and starts to establish a call with the collected number. The digit map pattern contains up to 52 options separated by a vertical bar (|) and enclosed in parenthesis. The maximum length of the entire digit pattern is limited to 152 characters. Available notations: • [n-m] represents a range of numbers • '.' (single dot) represents repetition • 'x' represents any single digit • 'T' represents a dial timer (configured by TimeBetweenDigits parameter) • 'S' should be used when a specific rule, that is part of a general rule, is to be applied immediately. For example, if you enter the general rule x.T and the specific rule 11x, you should append 'S' to the specific rule 11xS. For example: (11xS|00T|[1-7]xxx|8xxxxxxx|#xxxxxxx|*xx|91xxxxxxxxxx|9011x.T) **Note:** The digitmap mechanism is applicable only when ISDN Overlap dialing is used (ISDNRxOverlap = 1).
TimeBetweenDigits [Inter Digit Timeout for Overlap Dial]	Defines the time (in seconds) that the gateway waits between digits that are received from the ISDN when Tel→IP overlap dialing is performed. When this inter-digit timeout expires, the gateway uses the collected digits for the called destination number. The range is 1 to 10 seconds. The default value is 4 seconds.
MaxDigits [Max Digits In Phone Num for Overlap Dialing]	Defines the maximum number of collected destination number digits received from the ISDN when Tel→IP overlap dialing is performed. When the number of collected digits reaches the maximum, the gateway uses these digits for the called destination number. The range is 1 to 49. The default value is 30.
TimeForDialTone	Duration (in seconds) of the dial tone played to an ISDN terminal. Applicable to overlap dialing when 'ISDNInCallsBehavior = 65536'. The dial tone is played if the ISDN Setup message doesn't include the called number. The valid range is 0 to 60. The default time is 5 seconds.
SendMetering2IP [Send Metering Message to IP]	Enables or disables sending a metering tone INFO message to IP on detection of an MFC\R2 metering pulse. Valid options include: 0 = Disable (default). 1 = Enable.
R2Category [MFC R2 Category]	MFC R2 Calling Party Category (CPC). The parameter provides information on calling party such as National or International call, Operator or Subscriber and Subscriber priority. The parameter range is 1 to 15, defining one of the MFC R2 tones.
RegretTime	Determines the time period (in seconds) the gateway waits for an MFC R2 Resume (Reanswer) signal once a Suspend (Clear back) signal was received from the PBX. If this timer expires, the call is released. The valid range is 0 to 255. The default value is 0. Applicable only for MFC R2 CAS Brazil variant.
HeldTimeout	Determines the time period the gateway can stay on-hold. If a Resume (un-hold Re- INVITE) message is received before the timer expires, the call is renewed. If this timer expires, the call is released. -1 = Indefinitely (default). 0 - 2400=Time to wait in seconds. Currently applicable only to MFC R2 CAS variants.

Table 6-11: E1/T1/J1 Configuration Parameters (continues on pages 189 to 195)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
ISDN Flexible Behavior Parameters ISDN protocol is implemented in different Switches / PBXs by different vendors. Several implementations vary a little from the specification. Therefore, to provide a flexible interface that supports these ISDN variants, the ISDN behavior parameters are used.	
ISDNInCallsBehavior [Incoming Calls Behavior]	2048 = Sends Channel ID in the first response to an incoming Q.931 Call Setup message. Otherwise, the Channel ID is sent only if the gateway requires changing the proposed Channel ID (default). 8192 = Sends Channel ID in a Q.931 Call Proceeding message. 65536 = Includes Progress Indicator (PI=8) in Setup ACK message, if an empty called number is received in an incoming Setup message. Applicable to overlap dialing mode. The parameter also directs the gateway to play a dial tone (for 'TimeForDialTone'), until the next called number digits are received. 262144 = NI-2 second redirect number – Users can select and use (in INVITE messages) the NI-2 second redirect number, if two redirect numbers are received in Q.931 Setup for incoming Tel→IP calls. **Note:** To configure the gateway to support several 'ISDNInCallsBehavior' features, summarize the individual feature values. For example to support both '2048' and '65536' features, set 'ISDNInCallsBehavior = 67584.
ISDNIBehavior [Q.931 Layer Response Behavior]	1 = Q.931 Status message isn't sent if Q.931 received message contains an unknown/unrecognized IE(s). By default the Status message is sent. This parameter applies only to PRI variants in which sending of Status message is optional. 2 = Q.931 Status message isn't sent if an optional IE with invalid content is received. By default the Status message is sent. This parameter applies only to PRI variants in which sending of Status message is optional. 4 = Accepts unknown/unrecognized Facility IE. Otherwise, (default) the Q.931 message that contains the unknown Facility IE is rejected. This parameter applies to PRI variants where a complete ASN1 decoding is performed on Facility IE. 128 = Connect ACK message is sent in response to received Q.931 Connect. Applicable only to Euro ISDN User side outgoing calls. Otherwise, the Connect ACK is not sent (default). 512 = Enables to configure T1 NFAS Interface ID (refer to the parameter 'ISDNNFASInterfaceID_x'). Applicable to 4/5ESS, DMS, NI-2 and HKT variants. 2048 = Always set the Channel Identification IE to explicit Interface ID, even if the B-channel is on the same trunk as the D-channel. Applicable to 4/5ESS, DMS and NI-2 variants. 65536 = Applicable to ETSI, NI-2 and 5ESS. The calling party number (octet 3a) is always present even when presentation and screening are at their default. 131072 = Clears the call on reception of Q.931 Status with incompatible state. Otherwise, (default) no action is taken. **Note:** To configure the gateway to support several 'ISDNIBehavior' features, summarize the individual feature values. For example to support both '512' and '2048' features, set 'ISDNIBehavior = 2560'.

Table 6-11: E1/T1/J1 Configuration Parameters (continues on pages 189 to 195)

ini File Field Name Web Parameter Name	Valid Range and Description
ISDNGeneralCCBehavior [General Call Control Behavior]	16 = The gateway clears down the call if it receives a NOTIFY message specifying 'User-Suspended'. A NOTIFY (User-Suspended) message is used by some networks (e.g., in Italy or Denmark) to indicate that the remote user has cleared the call, especially in the case of a long distance voice call. 32 = Applies only to ETSI E1 lines (30B+D). Enables handling the differences between the newer QSIG standard (ETS 300-172) and other ETSI-based standards (ETS 300-102 and ETS 300-403) in the conversion of B-channel ID values into timeslot values: • In 'regular ETSI' standards, the timeslot is identical to the B-channel ID value, and the range for both is 1 to 15 and 17 to 31. The D-channel is identified as channel-id #16 and carried into the timeslot #16. • In newer QSIG standards, the channel-id range is 1 to 30, but the timeslot range is still 1 to 15 and 17 to 31. The D-channel is not identified as channel-id #16, but is still carried into the timeslot #16. When this bit is set, the channel ID #16 is considered as a valid B-channel ID, but timeslot values are converted to reflect the range 1 to 15 and 17 to 31. This is the new QSIG mode of operation. When this bit is not set (default), the channel_id #16 is not allowed, as for all ETSI-like standards.
ISDNOutCallsBehavior [Outgoing Calls Behavior]	1024 = Numbering plan / type for T1 IP→Tel calling number are defined according to the manipulation tables or according to RPID header (default). Otherwise, the Plan / type for T1 calls are set according to the length of the calling number
ISDNIBehavior_x [Q.931 Layer Response Behavior]	Same as the description for parameter 'ISDNIBehavior' for a specific trunk ID (x = 0 to 7)
ISDNInCallsBehavior_x [Incoming Calls Behavior]	Same as the description for parameter 'ISDNInCallsBehavior' for a specific trunk ID (x = 0 to 7)
ISDNOutCallsBehavior_x [Outgoing Calls Behavior]	Same as the description for parameter 'ISDNOutCallsBehavior' for a specific trunk ID (x = 0 to 7)

6.17 Channel Parameters

The Channel Parameters define the DTMF, fax and modem transfer modes. Refer to Section 8.3 on page 210 for a detailed description of Fax and Modem transfer modes; refer to Section 8.2 on page 208 for detailed description on DTMF transport modes.

Note that the Default Channel Parameters are applied to all the gateway's channels.

Table 6-12: Channel Parameters (continues on pages 196 to 200)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
DJBufMinDelay [Dynamic Jitter Buffer Minimum Delay]	0 to 150 msec (default = 10) Dynamic Jitter Buffer Minimum Delay. **Note:** For more information on the Jitter Buffer, refer to Section 8.6 on page 215.
DJBufOptFactor [Dynamic Jitter Buffer Optimization Factor]	Dynamic Jitter Buffer frame error / delay optimization factor. The valid range is 0 to 13. The default factor is 10. **Note 1:** Set to 13 for data (fax and modem) calls. **Note 2:** For more information on the Jitter Buffer, refer to Section 8.6 on page 215.
FaxTransportMode [Fax Transport Mode]	Fax transport mode that the gateway uses. You can select: 0 = Disable (transparent mode). 1 = T.38 Relay (default). 2 = Bypass. 3 = Events only. **Note:** If parameter IsFaxUsed = 1, then FaxTransportMode is always set to 1 (T.38 relay).
FaxRelayEnhancedRedundancy Depth [Fax Relay Enhanced Redundancy Depth]	Determines the number of repetitions applied to control packets when using T.38 standard. The valid range is 0 to 4. The default value is 0.
FaxRelayRedundancyDepth [Fax Relay Redundancy Depth]	Number of times that each fax relay payload is retransmitted to the network. The valid range is 0 to 2. The default value is 0.
FaxRelayMaxRate [Fax Relay Max Rate (bps)]	Limits the maximum rate at which fax messages are transmitted (outgoing calls). 0 = 2.4 kbps 1 = 4.8 kbps 2 = 7.2 kbps 3 = 9.6 kbps 4 = 12.0 kbps 5 = 14.4 kbps (default). **Note:** The rate is negotiated between the sides, i.e., the gateway adapts to the capabilities of the remote side.
FaxRelayECMEnable [Fax Relay ECM Enable]	0 = Disable using ECM (Error Correction Mode) mode during fax relay. 1 = Enable using ECM mode during fax relay (default).
FaxModemBypassCoderType [Fax/Modem Bypass Coder Type]	Coder the gateway uses when performing fax/modem bypass. Usually, high-bit-rate coders such as G.711 should be used. You can select: 0 = G.711 A-law 64 (default). 1 = G.711 μ-law.
CNGDetectorMode [CNG Detector Mode]	0 = Disable (default). 1 = Event Only (N/A). 2 = Relay. T.38 fax relay session is initiated by the originating fax if 'IsFaxUsed = 1'. Note that using this mode isn't recommended.

Table 6-12: Channel Parameters (continues on pages 196 to 200)

ini File Field Name Web Parameter Name	Valid Range and Description
FaxModemBypassM [Fax/Modem Bypass Packing Factor]	Number of (20 msec) coder payloads that are used to generate a fax/modem bypass packet. The valid range is 1, 2 or 3 coder payloads. The default value is 1 coder payload.
FaxBypassPayloadType [Fax Bypass Payload Type]	Determines the fax bypass RTP dynamic payload type. The valid range is 96 to 120. The default value is 102.
ModemBypassPayloadType	Modem Bypass dynamic payload type (range 0-127). The default value is 103.
DetFaxOnAnswerTone [Detect Fax on Answer Tone]	0 = Starts T.38 procedure on detection of V.21 preamble (default). 1 = Starts T.38 Procedure on detection of CED fax answering tone.
FaxModemBypassBasicRTPPacketInterval	0 = set internally (default) 1 = 5 msec (not recommended) 2 = 10 msec 3 = 20 msec
FaxModemBypassDJBufMinDelay	0 to 150 msec (default=40) Determines the Jitter Buffer delay during fax and modem bypass session
BellModemTransportType	Determines the Bell modem transport method. 0 = Transparent (default). 2 = Bypass. 3 = Transparent with events.
NSEMode	Cisco compatible fax and modem bypass mode 0 = NSE disabled (default) 1 = NSE enabled **Note 1:** This feature can be used only if VxxModemTransportType=2 (Bypass) **Note 2:** If NSE mode is enabled the SDP contains the following line: 'a=rtpmap:100 X-NSE/8000' **Note 3:** To use this feature: • The Cisco gateway must include the following definition: 'modem passthrough nse payload-type 100 codec g711alaw'. • Set the Modem transport type to Bypass mode ('VxxModemTransportType = 2') for all modems. • Configure the gateway parameter NSEPayloadType= 100 In NSE bypass mode the gateway starts using G.711 A-Law (default) or G.711µ-Law, according to the parameter 'FaxModemBypassCoderType'. The payload type used with these G.711 coders is a standard one (8 for G.711 A-Law and 0 for G.711 µ-Law). The parameters defining payload type for the 'old' AudioCodes' Bypass mode. 'FaxBypassPayloadType' and 'ModemBypassPayloadType' are not used with NSE Bypass. The bypass packet interval is selected according to the parameter 'FaxModemBypassBasicRtpPacketInterval'.
NSEPayloadType	NSE payload type for Cisco Bypass compatible mode. The valid range is 96-127. The default value is 105. **Note:** Cisco gateways usually use NSE payload type of 100.
V21ModemTransportType [V.21 Modem Transport Type]	V.21 Modem Transport Type that the gateway uses. You can select: 0 = Disable (Transparent) -- default 1 = Enable Relay – N/A 2 = Enable Bypass 3 = Events Only (Transparent with Events)

Table 6-12: Channel Parameters (continues on pages 196 to 200)

ini File Field Name Web Parameter Name	Valid Range and Description
V22ModemTransportType [V.22 Modem Transport Type]	V.22 Modem Transport Type that the gateway uses. You can select: 0 = Disable (Transparent) 1 = Enable Relay -- N/A 2 = Enable Bypass -- default 3 = Events Only (Transparent with Events)
V23ModemTransportType [V.23 Modem Transport Type]	V.23 Modem Transport Type that the gateway uses. You can select: 0 = Disable (Transparent) 1 = Enable Relay -- N/A 2 = Enable Bypass -- default 3 = Events Only (Transparent with Events)
V32ModemTransportType [V.32 Modem Transport Type]	V.32 Modem Transport Type that the gateway uses. You can select: 0 = Disable (Transparent) 1 = Enable Relay -- N/A 2 = Enable Bypass -- default 3 = Events Only (Transparent with Events) **Note:** This option applies to V.32 and V.32bis modems.
V34ModemTransportType [V.34 Modem Transport Type]	V.90 / V.34 Modem Transport Type that the gateway uses. You can select: 0 = Disable (Transparent) 1 = Enable Relay -- N/A 2 = Enable Bypass -- default 3 = Events Only (Transparent with Events)
InputGain [Input Gain]	PCM input gain control in dB. This parameter sets the level for the received (PSTN→IP) signal. The valid range is -32 to 31 dB. The default value is 0 dB. **Note:** This parameter is intended for advanced users. Changing it affects other gateway functionalities.
VoiceVolume [Voice Volume]	Voice gain control in dB. This parameter sets the level for the transmitted (IP→PSTN) signal. The valid range is -32 to 31 dB. The default value is 0 dB.
RTPRedundancyDepth [RTP Redundancy Depth]	0 = Disable redundancy packets generation (default) 1 = Enable generation of RFC 2198 redundancy packets.
RFC2198PayloadType	RTP redundancy packet payload type, according to RFC 2198. The range is 96-127. The default is 104. Applicable if 'RTPRedundancyDepth=1'
EnableSilenceCompression [Silence Suppression] The parameter **SCE** is used to maintain backward compatibility.	0 = Silence Suppression disabled (default). 1 = Silence Suppression enabled. 2 [Enable without adaptation] = A single silence packet is sent during silence period (applicable only to G.729). Silence Suppression is a method conserving bandwidth on VoIP calls by not sending packets when silence is detected. **Note:** If the selected coder is G.729, the following rules determine the value of the 'annexb' parameter of the fmtp attribute in the SDP. EnableSilenceCompression = 0 → 'annexb=no'. EnableSilenceCompression = 1 → 'annexb=yes'. EnableSilenceCompression = 2 and IsCiscoSCEMode = 0 → 'annexb=yes'. EnableSilenceCompression = 2 and IsCiscoSCEMode = 1 → 'annexb=no'.

Table 6-12: Channel Parameters (continues on pages 196 to 200)

ini File Field Name Web Parameter Name	Valid Range and Description
IsCiscoSCEMode	0 = There isn't a Cisco gateway at the remote side (default). 1 = There is a Cisco gateway at the remote side. When there is a Cisco gateway at the remote side, the local gateway must set the value of the 'annexb' parameter of the fmtp attribute in the SDP to 'no'. This logic should be used if 'EnableSilenceCompression = 2' (enable without adaptation). In this case, Silence Suppression should be used on the channel but not declared in the SDP.
EnableEchoCanceller [Echo Canceler] The parameter **ECE** is used to maintain backward compatibility.	0 = Echo Canceler disabled. 1 = Echo Canceler Enabled (default). **Note:** Refer also to the parameters 'MaxEchoCancellerLength' (described in Table 6-2 on page 138).
EnableNoiseReduction	Enables / disables the DSP Noise Reduction mechanism. 0 = Disable (default). 1 = Enable. **Note:** When this parameter is enabled the channel capacity might be reduced.
EnableStandardSIDPayloadType [Enable RFC 3389 CN Payload Type]	Determines whether Silence Indicator (SID) packets that are sent and received are according to RFC 3389. 0 = G.711 SID packets are sent in a proprietary method (default). 1 = SID (comfort noise) packets are sent with the RTP SID payload type according to RFC 3389. Applicable to G.711 and G.726 coders.
ComfortNoiseNegotiation [Comfort Noise Generation Negotiation]	Enables negotiation and usage of Comfort Noise (CN). Valid options include: 0 = Disable (default) 1 = Enable Comfort Noise negotiation The use of CN is indicated by including a payload type for CN on the media description line of the SDP. The agteway can use CN with a codec whose RTP timestamp clock rate is 8,000 Hz (G.711/G.726). The static payload type 13 is used. The use of CN is negotiated between sides; therefore, if the remote side doesn't support CN, it is not used. **Note:** Silence Suppression must be enabled to generate CN.
RTPSIDCoeffNum	Determines the number of spectral coefficients added to an SID packet being sent according to RFC 3389. Valid only if 'EnableStandardSIDPayloadType' is set to 1. The valid values are 0 (default), 4, 6, 8 and 10.
DTMFVolume [DTMF Volume]	DTMF gain control value in dB (to the TDM side). The valid range is -31 to 0 dB.The default value is -11 dB.
RxDTMFHangOverTime	Defines the Voice Silence time (in msec units) after playing DTMF or MF digits to the Tel / PSTN side that arrive as Relay from the IP side. Valid range is 0 to 2,000 msec. Default is 1,000 msec.
TxDTMFHangOverTime	Defines the Voice Silence time (in msec units) after detecting the end of DTMF or MF digits at the Tel / PSTN side when the DTMF Transport Type is either Relay or Mute. Valid range is 0 to 2,000 msec. Default is 100 msec.
DTMFTransportType [DTMF Transport Type]	0 = Erase digits from voice stream, do not relay to remote. 2 = Digits remain in voice stream. 3 = Erase digits from voice stream, relay to remote according to RFC 2833 (default). **Note:** This parameter is automatically updated if one of the following parameters is configured: TxDTMFOption or RxDTMFOption.

Table 6-12: Channel Parameters (continues on pages 196 to 200)

***ini* File Field Name** **Web Parameter Name**	**Valid Range and Description**
RFC2833PayloadType [RFC 2833 Payload Type]	The RFC 2833 DTMF relay dynamic payload type. Range: 96 to 99, 106 to 127; Default = 96 The 100, 102 to 105 range is allocated for proprietary usage. Cisco is using payload type 101 for RFC 2833. **Note:** When RFC 2833 payload type (PT) negotiation is used (TxDTMFOption=4), this payload type is used for the received DTMF packets. If negotiation isn't used, this payload type is used for receive and for transmit.
MGCPDTMFDetectionPoint	0 = DTMF event is reported on the end of a detected DTMF digit. 1 = DTMF event is reported on the start of a detected DTMF digit (default).
DTMFInterDigitInterval	Time in msec between generated DTMF digits to PSTN side (if TxDTMFOption = 1, 2 or 3). The default value is 100 msec. The valid range is 0 to 32767.
DTMFDigitLength	Time in msec for generating DTMF tones to the PSTN side (if TxDTMFOption = 1, 2 or 3). The default value is 100 msec. The valid range is 0 to 32767.
VQMonEnable [Enable RTCP XR]	Enables voice quality monitoring and RTCP Extended Reports (RTCP-XR). Valid options include: 0 = Disable (default) 1 = Enable For a description of the RTCP-XR reports, refer to Appendix F on page 367.
RTCPInterval [RTCP XR Packet Interval]	Defines the time interval (in msec) between adjacent RTCP reports. The interval range is 0 to 65,535. The default interval is 5,000.
DisableRTCPRandomize [Disable RTCP XR Interval Randomization]	Controls whether RTCP report intervals are randomized or whether each report interval accords exactly to the parameter RTCPInterval. Valid options include: 0 = Randomize (default) 1 = No Randomize
RTCPXREscIP [RTCP XR Collection Server]	IP address of the Event State Compositor (ESC). The gateway sends RTCP-XR reports using PUBLISH messages to this server. The address can be configured as a numerical IP address or as a domain name.
RTCPXRReportMode [RTCP XR Report Mode]	Determines whether or not RTCP-XR reports are sent to the Event State Compositor (ESC) and if so, defines the interval in which they are sent. Valid options include: 0 = Disable (RTCP-XR reports are not sent to the ESC) -- default 1 = End Call (RTCP-XR reports are sent to the ESC at the end of each call) 2 = End Call & Periodic (RTCP-XR reports are sent to the ESC at the end of each call and periodically according to the parameter RTCPInterval)

6.18 Configuration Files Parameters

The configuration files (Call Progress Tones, PRT, Voice Prompts and CAS) can be loaded to the gateway via the Embedded Web Server (refer to Section 5.8.2 on page 119), or via TFTP session.

➢ To load the configuration files via TFTP, take these 3 steps:

1. In the *ini* file, define the files to be loaded to the device. You can also define in the *ini* file whether the loaded files should be stored in the non-volatile memory so that the TFTP process is not required every time the device boots up.
2. Locate the configuration files you want to load and the *ini* file in the same directory.
3. Invoke a BootP/TFTP session; the *ini* and configuration files are loaded onto the device.

Table 6-13 below describes the *ini* file parameters that are associated with the configuration files.

Table 6-13: Configuration File Parameters

ini File Field Name	Valid Range and Description
CallProgressTonesFilename	The name of the file containing the Call Progress Tones definitions. Refer to Section 16 for additional information on how to create and load this file.
VoicePromptsFileName	The name (and path) of the file containing the Voice Prompts definitions. Refer to Section 16.2 on page 331 for additional information on how to create and load this file.
CASFileName	This is the name of the file containing specific CAS protocol definition (such as 'E_M_WinkTable.dat'). These files are provided to support various types of CAS signaling.
CASFileName_x	CAS file name (e.g., 'E_M_WinkTable.dat') that defines the CAS protocol. It is possible to define up to eight different CAS files by repeating this parameter. Each CAS file can be associated with one or more of the gateway trunks using the parameter CASTableIndex_x.
CASTablesNum	Number, 1 to 8. Specifies how many CAS configuration files are loaded.
PrerecordedTonesFileName	The name (and path) of the file containing the Prerecorded Tones.
UserInfoFileName	The name (and path) of the file containing the User Information data.
SaveConfiguration	Determines if the gateway's configuration (parameters and files) is saved to flash (non-volatile memory). 0 = Configuration isn't saved to flash memory. 1 = Configuration is saved to flash memory (default).

Reader's Notes

7 Using BootP / DHCP

The gateway uses the Bootstrap Protocol (BootP) and the Dynamic Host Configuration Protocol (DHCP) to obtain its networking parameters and configuration automatically after it is reset. BootP and DHCP are also used to provide the IP address of a TFTP server on the network, and files (*cmp* and *ini*) to be loaded into memory.

DHCP is a communication protocol that automatically assigns IP addresses from a central point. BootP is a protocol that enables a device to discover its own IP address. Both protocols have been extended to enable the configuration of additional parameters specific to the gateway.

> **Note:** BootP is normally used to initially configure the gateway. Thereafter, BootP is no longer required as all parameters can be stored in the gateway's non-volatile memory and used when BootP is inaccessible. BootP can be used again to change the IP address of the gateway (for example).

7.1 BootP/DHCP Server Parameters

BootP/DHCP can be used to provision the following parameters (included in the BootP/DHCP reply). Note that only the IP address and subnet mask are mandatory:

- **IP address, subnet mask:** Mandatory parameters that are sent to the gateway every time a BootP/DHCP process occurs.
- **Default gateway IP address:** An optional parameter that is sent to the gateway only if configured in the BootP/DHCP server.
- **TFTP server IP address:** An optional parameter that contains the address of the TFTP server from which the firmware (*cmp*) and *ini* files are loaded.
- **DNS server IP address (primary and secondary):** Optional parameters that contain the IP addresses of the primary and secondary DNS servers. These parameters are available only in DHCP and from Boot version 1.92.
- **Syslog server IP address:** An optional parameter that is sent to the gateway only if configured. This parameter is available only in DHCP.
- **SIP server IP address:** Two optional parameters that are sent to the gateway only if configured. These parameters are available only in DHCP.
- **Firmware file name**: An optional parameter that contains the name of the firmware file to be loaded to the gateway via TFTP.
- ***ini* file name:** An optional parameter that contains the name of the *ini* file to be loaded to the gateway via TFTP.

7.2 Using DHCP

When the gateway is configured to use DHCP (DHCPEnable = 1), it attempts to contact the enterprise's DHCP server to obtain the networking parameters (IP address, subnet mask, default gateway, primary/secondary DNS server and two SIP server addresses). These network parameters have a 'time limit'. After the time limit expires, the gateway must 'renew' its lease from the DHCP server.

Note that if the DHCP server denies the use of the gateway's current IP address and specifies a different IP address (according to RFC 1541), the gateway must change its networking parameters. If this happens while calls are in progress, they are not automatically rerouted to the new network address (since this function is beyond the scope of a VoIP gateway). Therefore, administrators are advised to configure DHCP servers to allow renewal of IP addresses.

Note: If the gateway's network cable is disconnected and reconnected, a DHCP renewal is performed (to verify that the gateway is still connected to the same network).

When DHCP is enabled, the gateway also includes its product name (e.g., 'Mediant 2000') in the DHCP 'option 60' Vendor Class Identifier. The DHCP server can use this product name to assign an IP address accordingly.

Note: After power-up, the gateway performs two distinct DHCP sequences. Only in the second sequence, DHCP 'option 60' is contained. If the gateway is reset from the Web/SNMP, only a single DHCP sequence containing 'option 60' is sent.

If DHCP procedure is used, the new gateway IP address, allocated by the DHCP server, must be detected.

Note: If, during operation, the IP address of the gateway is changed as a result of a DHCP renewal, the gateway is automatically reset.

➢ To detect the gateway's IP address, follow one of the procedures below:

- Starting with Boot version 1.92, the gateway can use a host name in the DHCP request. The host name is set to acl_nnnnn, where nnnnn stands for the gateway's serial number (the serial number is equal to the last 6 digits of the MAC address converted from Hex to decimal). If the DHCP server registers this host name to a DNS server, the user can access the gateway (through a Web browser) using a URL of http://acl_<serial number> (instead of using the gateway's IP address). For example, if the gateway's MAC address is 00908f010280, the DNS name is acl_66176.
- After physically resetting the gateway its IP address is displayed in the 'Client Info' column in the BootP/TFTP configuration utility (refer to Figure D-1 on page 355).
- Contact your System Administrator.

7.3 Using BootP

7.3.1 Upgrading the Gateway

When upgrading the gateway (loading new software onto the gateway) using the BootP/TFTP configuration utility:

- From version 4.4 to version 4.4 or to any higher version, the device retains its configuration (*ini* file). However, the auxiliary files (CPT, logo, etc.) may be erased.
- From version 4.6 to version 4.6 or to any higher version, the device retains its configuration (*ini* file) and auxiliary files (CPT, logo, etc.).

You can also use the Software Upgrade wizard, available through the Web Interface (refer to Section 5.8.1 on page 115).

Note: To save the *cmp* file to non-volatile memory, use the -fb command line switches. If the file is not saved, the gateway reverts to the old version of software after the next reset. For information on using command line switches, refer to Section D.11.6 on page 362.

7.3.2 Vendor Specific Information Field

The gateway uses the vendor specific information field in the BootP request to provide device-related initial startup information. The BootP/TFTP configuration utility displays this information in the 'Client Info' column (refer to Figure D-1).

Note: This option is not available on DHCP servers.

The Vendor Specific Information field is disabled by default. To enable / disable this feature: set the *ini* file parameter 'ExtBootPReqEnable' (Table 6-2 on page 138) or use the '-be' command line switch (refer to Table D-1 on page 363).

Table 7-1 details the vendor specific information field according to device types:

Table 7-1: Vendor Specific Information Field

Tag #	**Description**	**Value**	**Length**
220	Gateway Type	#02 = TP-1610 #05 = TP-260	1
221	Current IP Address	XXX.XXX.XXX.XXX	4
222	Burned Boot Software Version	X.XX	4
223	Burned *cmp* Software Version	XXXXXXXXXXXX	12
224	Geographical Address	0 – 31 (TP-260 Only)	1
225	Chassis Geographical Address	0 – 31 (TP-260 Only)	1
229	E&M	N/A	1

Table 7-2 exemplifies the structure of the vendor specific information field for a TP-1610 slave module with IP address 10.2.70.1.

Table 7-2: Structure of the Vendor Specific Information Field

Vendor-Specific Information Code	Total Length	Tag Num	Length	Value	Tab Num	Length	Value	Tag Num	Length	Value (1)	Value (2)	Value (3)	Value (4)	Tag End
42	12	220	1	2	225	1	1	221	4	10	2	70	1	255

8 Telephony Capabilities

8.1 Working with Supplementary Services

The gateway gateway supports the following supplementary services:

- Call Hold / Retrieve (refer to Section 8.1.1 on page 207)
- Call Transfer (refer to Section 8.1.2 on page 207)
- Call Forward (doesn't initiate call forward, only responds to call forward request)
- Call Waiting

The gateway SIP users are only required to enable the Hold and Transfer features. The call forward (supporting 30x redirecting responses) and call waiting (receive of 182 response) features are enabled by default. Note that all call participants must support the specific used method.

Note: When working with application servers (such as BroadSoft's BroadWorks) in client server mode (the application server controls all supplementary services and keypad features by itself), the gateway's supplementary services must be disabled.

8.1.1 Call Hold and Retrieve Features

- The party that initiates the hold is called the *holding* party, the other party is called the *held* party. The gateway can't initiate the hold, but it can respond to hold request, and as such it is a held party.
- After a successful hold, the held party should hear HELD_TONE, defined in the gateway's Call Progress Tones file.
- Retrieve can be performed only by the holding party while the call is held and active.
- After a successful retrieve the voice should be connected again.
- The hold and retrieve functionalities are implemented by Reinvite messages. The IP address 0.0.0.0 as the connection IP address or the string 'a=inactive' in the received Reinvite SDP cause the gateway to enter Hold state and to play held tone (configured in the gateway) to the PBX/PSTN. If the string 'a=recvonly' is received in the SDP message, the gateway stops sending RTP packets, but continues to listen to the incoming RTP packets. Usually, the remote party plays, in this scenario, Music on Hold (MOH) and the gateway forwards the MOH to the held party.

8.1.2 Call Transfer

There are two types of call transfers:

- Consultation Transfer
- Blind Transfer

The common way to perform a consultation transfer is as follows:

In the transfer scenario there are three parties:

Party A - transferring, Party B – transferred, Party C – transferred to.

- A Calls B.
- B answers.
- A presses the hookflash and puts B on-hold (party B hears a hold tone).
- A dials C.

- After A completed dialing C, A can perform the transfer by on-hooking the A phone.
- After the transfer is completed, B and C parties are engaged in a call.

The transfer can be initiated at any of the following stages of the call between A to C:

- Just after completing dialing C phone number - Transfer from setup.
- While hearing ring back – Transfer from alert.
- While speaking to C – Transfer from active.

Blind transfer is performed after we have a call between A and B, and A party decides to transfer the call to C immediately without speaking with C.

The result of the transfer is a call between B and C (just like consultation transfer only skipping the consultation stage).

The gateway doesn't initiate call transfer, it only can respond to call transfer request.

8.2 Configuring the DTMF Transport Types

You can control the way DTMF digits are transported over the IP network to the remote endpoint. The following five modes are supported:

1. Using INFO message according to the Nortel IETF draft:
   In this mode DTMF digits are carried to the remote side within INFO messages.
   To enable this mode set:
   - RxDTMFOption = 0
     (**Protocol Management** menu > **Protocol Definition** submenu > **DTMF & Dialing** option > 'Declare RFC 2833 in SDP' = No)
   - TxDTMFOption = 1
     (**Protocol Management** menu > **Protocol Definition** submenu > **DTMF & Dialing** option > '1st to 5th Tx DTMF Option' = INFO (Nortel))

   Note that in this mode, DTMF digits are erased from the audio stream (DTMFTransportType is automatically set to 0 (DTMF Mute)).
2. Using INFO message according to Cisco's style:
   In this mode DTMF, digits are carried to the remote side within INFO messages.
   To enable this mode set:
   - RxDTMFOption = 0 ('Declare RFC 2833 in SDP' = No)
   - TxDTMFOption = 3 ('1st to 5th Tx DTMF Option' = INFO (Cisco))

   Note that in this mode, DTMF digits are erased from the audio stream (DTMFTransportType is automatically set to 0 (DTMF Mute)).
3. Using NOTIFY messages according to <draft-mahy-sipping-signaled-digits-01.txt>:
   In this mode, DTMF digits are carried to the remote side using NOTIFY messages.
   To enable this mode set:
   - RxDTMFOption = 0 ('Declare RFC 2833 in SDP' = No)
   - TxDTMFOption = 2 ('1st to 5th DTMF Option' = NOTIFY)

   Note that in this mode, DTMF digits are erased from the audio stream (DTMFTransportType is automatically set to 0 (DTMF Mute)).

4. Using RFC 2833 relay with Payload type negotiation:
   In this mode, DTMF digits are carried to the remote side as part of the RTP stream in accordance with RFC 2833 standard.
   To enable this mode set:
   - TxDTMFOption = 4 ('1st to 5th Tx DTMF Option' = RFC 2833)
   - RxDTMFOption = 3 ('Declare RFC 2833 in SDP' = Yes)

   Note that to set the RFC 2833 payload type with a different value (other than its default, 96) configure the 'RFC2833PayloadType' (RFC 2833 Payload Type) parameter. The gateway negotiates the RFC 2833 payload type using local and remote SDP and sends packets using the PT from the received SDP. The gateway expects to receive RFC 2833 packets with the same PT as configured by the 'RFC2833PayloadType' parameter. If the remote side doesn't include 'telephony-event' in its SDP, the gateway sends DTMF digits in transparent mode (as part of the voice stream).
5. Sending DTMF digits (in RTP packets) as part of the audio stream (DTMF Relay is disabled):
   Note that this method is normally used with G.711 coders; with other Low Bit Rate (LBR) coders the quality of the DTMF digits is reduced.
   To set this mode:
   - TxDTMFOption = 0 ('1st to 5th Tx DTMF Option' = Disable)
   - RxDTMFOption = 0 ('Declare RFC 2833 in SDP' = No)
   - DTMFTransportType = 2 ('DTMF Transport Type' = Transparent DTMF)

Note 1: The gateway is always ready to receive DTMF packets over IP, in all possible transport modes: INFO messages, NOTIFY, and RFC 2833 (in proper payload type) or as part of the audio stream.

Note 2: To exclude RFC 2833 Telephony event parameter from the gateway's SDP, set RxDTMFOption = 0 in the *ini* file.

The following parameters affect the way the gateway handles the DTMF digits:

- TxDTMFOption and RxDTMFOption (described in Table 6-7).
- RFC2833PayloadType, MGCPDTMFDetectionPoint, DTMFDigitLength, DTMFVolume, DTMFInterDigitInterval and DTMFTransportType (described in Table 6-12).

8.3 Fax & Modem Transport Modes

8.3.1 Fax/Modem Settings

Users can choose to use for fax, and for each modem type (V.22/V.23/Bell/V.32/V.34), one of the following transport methods:

- Fax relay mode (demodulation / remodulation, not applicable to Modem),
- Bypass (using a high bit rate coder to pass the signal), or
- Transparent (passing the signal in the current voice coder).

When any of the relay modes are enabled, distinction between fax and modem is not immediately possible at the beginning of a session. The channel is therefore in 'Answer Tone' mode until a decision is made. The packets sent to the network at this stage are T.38-complaint fax relay packets.

8.3.1.1 Configuring Fax Relay Mode

When FaxTransportMode = 1 (relay mode), and when fax is detected, the channel automatically switches from the current voice coder to answer tone mode, and then to T.38-complaint fax relay mode.

When fax transmission ends, the reverse is carried out, and fax relay switches to voice. This mode switch occurs automatically, both at the local and remote endpoints.

Users can limit the fax rate using the FaxRelayMaxRate parameter and can enable/disable ECM fax mode using the FaxRelayECMEnable parameter.

When using T.38 mode, the user can define a redundancy feature to improve fax transmission over congested IP network. This feature is activated by 'FaxRelayRedundancyDepth' and 'FaxRelayEnhancedRedundancyDepth' parameters. Although this is a proprietary redundancy scheme, it should not create problems when working with other T.38 decoders.

Note: T.38 mode currently supports only the T.38 UDP syntax.

8.3.1.2 Configuring Fax/Modem ByPass Mode

When VxxTransportType=2 (FaxModemBypass, Vxx can be either V32/V22/Bell/V34/Fax), then when fax/modem is detected, the channel automatically switches from the current voice coder to a high bit-rate coder, as defined by the user, with the FaxModemBypassCoderType configuration parameter.

During the bypass period, the coder uses the packing factor (by which a number of basic coder frames are combined together in the outgoing WAN packet) set by the user in the FaxModemBypassM configuration parameter. The network packets to be generated and received during the bypass period are regular voice RTP packets (per the selected bypass coder) but with a different RTP Payload type.

When fax/modem transmission ends, the reverse is carried out, and bypass coder is switched to regular voice coder.

8.3.1.3 Supporting V.34 Faxes

V.34 fax machine support is available only in bypass mode (fax relay is not supported) when the channel is configured in one of the configurations described below:

```
FaxTransportMode = 2 (Bypass)
V34ModemTransportType = 2 (Modem bypass)
```

In this configuration, both T.30 and V.34 faxes work in Bypass mode

Or

```
FaxTransportMode = 1 (Relay)
V34ModemTransportType = 2 (Modem bypass)
```

In this configuration, T.30 faxes use Relay mode (T.38) while V.34 fax uses Bypass mode.

In order to use V.34 fax in Relay mode (T.38), you must configure:

```
FaxTransportMode = 1 (Relay)
V34ModemTransportType = 0 (Transparent)
V32ModemTransportType = 0
V23ModemTransportType = 0
V22ModemTransportType = 0
```

This configuration forces the V.34 fax machine to work in T.30 mode.

8.4 Event Notification using X-Detect Header

The gateway supports the sending of notifications to a remote party notifying the occurrence (or detection) of certain events on the media stream. Event detection and notifications is performed using the X-Detect SIP message header, and only when establishing a SIP dialog.

For supporting some events, certain gateway configurations need to be performed. The table below lists the support event types (and subtypes) and the corresponding gateway configurations, if required:

Table 8-1: Supported X-Detect Event Types

Events		Required Gateway Configuration
Type	**Subtype**	
AMD	voice automata silence unknown	EnableDSPIPMDetectors = 1 AMDTimeout = 2000 (msec)
CPT	SIT	SITDetectorEnable = 1 UserDefinedToneDetectorEnable = 1 **Note:** Differentiation of SIT is not supported in 5.0.
FAX	CED	(IsFaxUsed ≠ 0) or (IsFaxUsed = 0 and FaxTransportMode ≠ 0)
	modem	VxxModemTransportType = 3
PTT	voice-start voice-end	EnableDSPIPMDetectors = 1

The X-Detect event notification process is as follows:

1. For IP-to-Tel or Tel-to-IP calls, the gateway receives a SIP request message (using the X-Detect header) that the remote party wishes to detect events on the media stream. For incoming (IP-to-Tel) calls, the request must be indicated in the initial INVITE and responded to either in the 183 response (for early dialogs) or in the 200 OK response (for confirmed dialogs). For outgoing calls (Tel-to-IP), the request may be received in the 183 (for early dialogs) and responded to in the PRACK, or received in the 200 OK (for confirmed dialogs) and responded to in the ACK.
2. Once the gateway receives such a request, it sends a SIP response message (using the X-Detect header) to the remote party, listing all supported events that can be detected. The absence of the X-Detect header indicates that no detections are available.
3. Each time the gateway detects a supported event, the event is notified to the remote party, by sending an INFO message with the following message body:
   - Content-Type: application/X-DETECT
   - Type = [AMD | CPT | FAX | PTT…]
   - Subtype = xxx (according to the defined subtypes of each type)

Below is an example of SIP messages implementing the X-Detect header:

```
INVITE sip:101@10.33.2.53;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.33.2.53;branch=z9hG4bKac5906
Max-Forwards: 70
From: "anonymous" <sip:anonymous@anonymous.invalid>;tag=1c25298
To: <sip:101@10.33.2.53;user=phone>
Call-ID: 11923@10.33.2.53
CSeq: 1 INVITE
Contact: <sip:100@10.33.2.53>
X- Detect: Request=CPT,FAX

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.33.2.53;branch=z9hG4bKac5906
From: "anonymous" <sip:anonymous@anonymous.invalid>;tag=1c25298
To: <sip:101@10.33.2.53;user=phone>;tag=1c19282
Call-ID: 11923@10.33.2.53
CSeq: 1 INVITE
Contact: <sip:101@10.33.2.53>
X- Detect: Response=CPT,FAX

INFO sip:101@10.33.2.53;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.33.2.53;branch=z9hG4bKac5906
Max-Forwards: 70
From: "anonymous" <sip:anonymous@anonymous.invalid>;tag=1c25298
To: <sip:101@10.33.2.53;user=phone>
Call-ID: 11923@10.33.2.53
CSeq: 1 INVITE
Contact: <sip:100@10.33.2.53>
X- Detect: Response=CPT,FAX
Content-Type: Application/X-Detect
Content-Length: xxx
Type = CPT
Subtype = SIT
```

8.5 ThroughPacket™

The gateway supports a proprietary method to aggregate RTP streams from several channels to reduce the bandwidth overhead caused by the attached Ethernet, IP, UDP and RTP headers, and to reduce the packet / data transmission rate. This option reduces the load on network routers and can typically save 50% (e.g., for G.723) on IP bandwidth.

ThroughPacket™ is accomplished by aggregating payloads from several channels that are sent to the same destination IP address into a single IP packet.

ThroughPacket™ can be applied to the entire gateway or, using IP Profile, to specific IP destinations (refer to Section 5.5.6.3 on page 80). Note that ThroughPacket™ must be enabled on both gateways.

To enable ThroughPacket™ set the parameter 'RemoteBaseUDPPort' to a nonzero value. Note that the value of 'RemoteBaseUDPPort' on the local gateway must equal the value of 'BaseUDPPort' of the remote gateway. The gateway uses these parameters to identify and distribute the payloads from the received multiplexed IP packet to the relevant channels.

In ThroughPacket™ mode, the gateway uses a single UDP port for all incoming multiplexed packets and a different port for outgoing packets. These ports are configured using the parameters 'L1L1ComplexTxUDPPort' and 'L1L1ComplexRxUDPPort'.

When ThroughPacket™ is used, Call statistics aren't available (since there is no RTCP flow).

8.6 Dynamic Jitter Buffer Operation

Voice frames are transmitted at a fixed rate. If the frames arrive at the other end at the same rate, voice quality is perceived as good. In many cases, however, some frames can arrive slightly faster or slower than the other frames. This is called jitter (delay variation), and degrades the perceived voice quality. To minimize this problem, the gateway uses a jitter buffer. The jitter buffer collects voice packets, stores them and sends them to the voice processor in evenly spaced intervals.

The gateway uses a dynamic jitter buffer that can be configured using two parameters:

- Minimum delay, 'DJBufMinDelay' (0 msec to 150 msec). Defines the starting jitter capacity of the buffer. For example, at 0 msec, there is no buffering at the start. At the default level of 10 msec, the gateway always buffers incoming packets by at least 10 msec worth of voice frames.
- Optimization Factor, 'DJBufOptFactor' (0 to 12, 13). Defines how the jitter buffer tracks to changing network conditions. When set at its maximum value of 12, the dynamic buffer aggressively tracks changes in delay (based on packet loss statistics) to increase the size of the buffer and doesn't decays back down. This results in the best packet error performance, but at the cost of extra delay. At the minimum value of 0, the buffer tracks delays only to compensate for clock drift and quickly decays back to the minimum level. This optimizes the delay performance but at the expense of a higher error rate.

The default settings of 10 msec Minimum delay and 10 Optimization Factor should provide a good compromise between delay and error rate. The jitter buffer 'holds' incoming packets for 10 msec before making them available for decoding into voice. The coder polls frames from the buffer at regular intervals to produce continuous speech. As long as delays in the network do not change (jitter) by more than 10 msec from one packet to the next, there is always a sample in the buffer for the coder to use. If there is more than 10 msec of delay at any time during the call, the packet arrives too late. The coder tries to access a frame and is not able to find one. The coder must produce a voice sample even if a frame is not available. It therefore compensates for the missing packet by adding a Bad-Frame-Interpolation (BFI) packet. This loss is then flagged as the buffer being too small. The dynamic algorithm then causes the size of the buffer to increase for the next voice session. The size of the buffer may decrease again if the gateway notices that the buffer is not filling up as much as expected. At no time does the buffer decrease to less than the minimum size configured by the Minimum delay parameter.

Special Optimization Factor Value: 13

One of the purposes of the Jitter Buffer mechanism is to compensate for clock drift. If the two sides of the VoIP call are not synchronized to the same clock source, one RTP source generates packets at a lower rate, causing under-runs at the remote Jitter Buffer. In normal operation (optimization factor 0 to 12), the Jitter Buffer mechanism detects and compensates for the clock drift by occasionally dropping a voice packet or by adding a BFI packet.

Fax and modem devices are sensitive to small packet losses or to added BFI packets. Therefore to achieve better performance during modem and fax calls, the Optimization Factor should be set to 13. In this special mode the clock drift correction is performed less frequently - only when the Jitter Buffer is completely empty or completely full. When such condition occurs, the correction is performed by dropping several voice packets simultaneously or by adding several BFI packets simultaneously, so that the Jitter Buffer returns to its normal condition.

8.7 Configuring the Gateway's Alternative Routing (based on Connectivity and QoS)

The Alternative Routing feature enables reliable routing of Tel to IP calls when Proxy isn't used. The gateway periodically checks the availability of connectivity and suitable Quality of Service (QoS) before routing. If the expected quality cannot be achieved, an alternative IP route for the prefix (phone number) is selected.

Note that if the alternative routing destination is the gateway itself, the call can be configured to be routed back to one of the gateway's trunk groups and thus back into the PSTN (PSTN Fallback).

8.7.1 Alternative Routing Mechanism

When a Tel→IP call is routed through the gateway, the call's destination number is compared to the list of prefixes defined in the Tel to IP Routing table (described in Section 5.5.5.1 on page 70). The Tel to IP Routing table is scanned for the destination number's prefix starting at the top of the table. When an appropriate entry (destination number matches one of the prefixes) is found; the prefix's corresponding destination IP address is checked. If the destination IP address is disallowed, an alternative route is searched for in the following table entries.

Destination IP address is disallowed if no ping to the destination is available (ping is continuously initiated every 7 seconds), when an inappropriate level of QoS was detected, or when DNS host name is not resolved. The QoS level is calculated according to delay or packet loss of previously ended calls. If no call statistics are received for two minutes, the QoS information is reset.

The gateway matches the rules starting at the top of the table. For this reason, enter the main IP route above any alternative route.

8.7.2 Determining the Availability of Destination IP Addresses

To determine the availability of each destination IP address (or host name) in the routing table, one (or all) of the following (configurable) methods are applied:

- **Connectivity:** The destination IP address is queried periodically (currently only by ping).
- **QoS:** The QoS of an IP connection is determined according to RTCP (Real-Time Control Protocol) statistics of previous calls. Network delay (in msec) and network packet loss (in percentage) are separately quantified and compared to a certain (configurable) threshold. If the calculated amounts (of delay or packet loss) exceed these thresholds the IP connection is disallowed.
- **DNS resolution:** When host name is used (instead of IP address) for the destination route, it is resolved to an IP address by a DNS server. Connectivity and QoS are then applied to the resolved IP address.

8.7.3 PSTN Fallback as a Special Case of Alternative Routing

The purpose of the PSTN Fallback feature is to enable the gateway to redirect PSTN originated calls back to the legacy PSTN network if a destination IP route is found unsuitable (disallowed) for voice traffic at a specific time.

To enable PSTN fallback, assign the IP address of the gateway itself as an alternative route to the desired prefixes. Note that calls (now referred to as IP to Tel calls) can be re-routed to a specific trunk group using the Routing parameters.

8.7.4 Relevant Parameters

The following parameters (described in Table 6-10) are used to configure the Alternative Routing mechanism:

- AltRoutingTel2IPEnable
- AltRoutingTel2IPMode
- IPConnQoSMaxAllowedPL
- IPConnQoSMaxAllowedDelay

8.8 Call Detail Report

The Call Detail Report (CDR) contains vital statistic information on calls made by the gateway. CDRs are generated at the end and (optionally) at the beginning of each call (determined by the parameter 'CDRReportLevel'). The destination IP address for CDR logs is determined by the parameter 'CDRSyslogServerIP'.

The following CDR fields are supported:

Table 8-2: Supported CDR Fields (continues on pages 217 to 218)

Field Name	Description
Cid	Board's Logic Channel Number
CallId	H.323/SIP Call Identifier
Trunk	Physical Trunk Number
BChan	Selected B-Channel
ConId	H.323/SIP Conference ID
TG	Trunk Group Number
EPTyp	Endpoint Type
Orig	Call Originator (IP, Tel)
SourceIp	Source IP Address
DestIp	Destination IP Address
TON	Source Phone Number Type
NPI	Source Phone Number Plan
SrcPhoneNum	Source Phone Number
SrcNumBeforeMap	Source Number Before Manipulation
TON	Destination Phone Number Type
NPI	Destination Phone Number Plan
DstPhoneNum	Destination Phone Number
DstNumBeforeMap	Destination Number Before Manipulation
Durat	Call Duration
Coder	Selected Coder
Intrv	Packet Interval
RtpIp	RTP IP Address
Port	Remote RTP Port
TrmSd	Initiator of Call Release (IP, Tel, Unknown)
TrmReason	Termination Reason
Fax	Fax Transaction during the Call
InPackets	Number of Incoming Packets
OutPackets	Number of Outgoing Packets
PackLoss	Number of Incoming Lost Packets
UniqueId	unique RTP ID
SetupTime	Call Setup Time
ConnectTime	Call Connect Time

Table 8-2: Supported CDR Fields (continues on pages 217 to 218)

Field Name	Description
ReleaseTime	Call Release Time
RTPdelay	RTP Delay
RTPjitter	RTP Jitter
RTPssrc	Local RTP SSRC
RemoteRTPssrc	Remote RTP SSRC
RedirectReason	Redirect Reason
TON	Redirection Phone Number Type
NPI	Redirection Phone Number Plan
RedirectPhonNum	Redirection Phone Number

8.9 Supported RADIUS Attributes

Use Table 8-3 below for explanations on the RADIUS attributes contained in the communication packets transmitted between the gateway and a RADIUS Server.

Table 8-3: Supported RADIUS Attributes (continues on pages 218 to 219)

Attribute Number	Attribute Name	VSA No.	Purpose	Value Format	Sample	AAA [1]
Request Attributes						
1	User-Name		Account number or calling party number or blank	String up to 15 digits long	5421385747	Start Acc Stop Acc
4	NAS-IP-Address		IP address of the requesting AudioCodes gateway	Numeric	192.168.14.43	Start Acc Stop Acc
6	Service-Type		Type of service requested	Numeric	1: login	Start Acc Stop Acc
26	h323-incoming-conf-id	1	H.323/SIP call identifier	Up to 32 octets		Start Acc Stop Acc
26	h323-remote-address	23	IP address of the remote gateway	Numeric		Stop Acc
26	h323-conf-id	24	H.323/SIP call identifier	Up to 32 octets		Start Acc Stop Acc
26	h323-setup-time	25	Setup time in NTP format 1	String		Start Acc Stop Acc
26	h323-call-origin	26	The call's originator: Answering (IP) or Originator (PSTN)	String	Answer, Originate etc	Start Acc Stop Acc
26	h323-call-type	27	Protocol type or family used on this leg of the call	String	VoIP	Start Acc Stop Acc
26	h323-connect-time	28	Connect time in NTP format	String		Stop Acc
26	h323-disconnect-time	29	Disconnect time in NTP format	String		Stop Acc
26	h323-disconnect-cause	30	Q.931 disconnect cause code	Numeric		Stop Acc

[1] The values in column 'AAA' are as follows:
'Start Acc' - Start Accounting
'Stop Acc' - Stop Accounting

Table 8-3: Supported RADIUS Attributes (continues on pages 218 to 219)

Attribute Number	Attribute Name	VSA No.	Purpose	Value Format	Sample	AAA [1]
26	h323-gw-id	33	Name of the gateway	String	SIPIDString	Start Acc Stop Acc
30	Called-Station-Id			String	8004567145	Start Acc
			Destination phone number	String	2427456425	Stop Acc
31	Calling-Station-Id		Calling Party Number (ANI)	String	5135672127	Start Acc Stop Acc
40	Acct-Status-Type		Account Request Type (start or stop) **Note:** 'start' isn't supported on the Calling Card application.	Numeric	1: start, 2: stop	Start Acc Stop Acc
41	Acct-Delay-Time		No. of seconds tried in sending a particular record	Numeric	5	Start Acc Stop Acc
42	Acct-Input-Octets		Number of octets received for that call duration	Numeric		Stop Acc
43	Acct-Output-Octets		Number of octets sent for that call duration	Numeric		Stop Acc
44	Acct-Session-Id		A unique accounting identifier - match start & stop	String	34832	Start Acc Stop Acc
46	Acct-Session-Time		For how many seconds the user received the service	Numeric		Stop Acc
47	Acct-Input-Packets		Number of packets received during the call	Numeric		Stop Acc
48	Acct-Output-Packets		Number of packets sent during the call	Numeric		Stop Acc
61	NAS-Port-Type		Gateway's physical port type on which the call is active	String	0: Asynchronous	Start Acc Stop Acc
Response Attributes						
26	h323-return-code	103	The reason for failing authentication (0 = ok, other number failed)	Numeric	0 Request accepted	Stop Acc
44	Acct-Session-Id		A unique accounting identifier – match start & stop	String		Stop Acc

8.9.1 RADIUS Server Messages

In Figure 8-1 below, non-standard parameters are preceded with brackets.

Figure 8-1: Accounting Example

```
Accounting-Request (361)
user-name = 111
acct-session-id = 1
nas-ip-address = 212.179.22.213
nas-port-type = 0
acct-status-type = 2
acct-input-octets = 4841
acct-output-octets = 8800
acct-session-time = 1
acct-input-packets = 122
acct-output-packets = 220
called-station-id = 201
calling-station-id = 202
// Accounting non-standard parameters:
(4923 33) h323-gw-id =
(4923 23) h323-remote-address = 212.179.22.214
(4923 1) h323-ivr-out = h323-incoming-conf-id:02102944 600a1899 3fd61009 0e2f3cc5
(4923 30) h323-disconnect-cause = 22 (0x16)
(4923 27) h323-call-type = VOIP
(4923 26) h323-call-origin = Originate
(4923 24) h323-conf-id = 02102944 600a1899 3fd61009 0e2f3cc5
```

8.10 Trunk to Trunk Routing Example

This example describes two AudioCodes gateways, each interface with the PSTN through four E1 spans. Gateway 'A' is configured to route all incoming Tel→IP calls to gateway 'B'. Gateway 'B' generates calls to PSTN on the same E1 Trunk as the call was originally received (in gateway 'A').

- Gateway 'A' IP address is 192.168.3.50
- Gateway 'B' IP address is 192.168.3.51

***Ini* File Parameters of Gateways 'A' and 'B':**

1. Define, for both gateways, four trunk groups; each with 30 B-channels:
   - TrunkGroup_1 = 0/1-31,1000
   - TrunkGroup_2 = 1/1-31,2000
   - TrunkGroup_3 = 2/1-31,3000
   - TrunkGroup_4 = 3/1-31,4000
2. In gateway 'A', add the originating Trunk Group ID, as a prefix, to the destination number, for Tel→IP calls:
   AddTrunkGroupAsPrefix=1
3. In gateway 'A', route all incoming PSTN calls, starting with the prefixes 1, 2, 3 and 4, to gateway's 'B' IP address:
   - Prefix = 1, 192.168.3.51
   - Prefix = 2, 192.168.3.51
   - Prefix = 3, 192.168.3.51
   - Prefix = 4, 192.168.3.51

   Note: It is also possible to define 'Prefix = *,192.168.3.51' instead of the four lines above.
4. In gateway 'B', route IP→PSTN calls to Trunk Group ID according to the first digit of the called number:
   - PSTNPrefix = 1,1
   - PSTNPrefix = 2,2
   - PSTNPrefix = 3,4
   - PSTNPrefix = 4,4
5. In gateway 'B', remove the first digit from each IP→PSTN number, before it is used in an outgoing call:
   NumberMapIP2Tel = *,1

8.11 Proxy or Registrar Registration Example

```
REGISTER sip:servername SIP/2.0
VIA: SIP/2.0/UDP 212.179.22.229;branch=z9hG4bRaC7AU234
From: <sip:GWRegistrationName@sipgatewayname>;tag=1c29347
To: <sip:GWRegistrationName@sipgatewayname>
Call-ID: mailto:10453@212.179.22.229
Seq: 1 REGISTER
Expires: 3600
Contact: sip:GWRegistrationName@212.179.22.229
Content-Length: 0
```

The ‘servername’ string is defined according to the following rules:

- The ‘servername’ is equal to ‘RegistrarName’ if configured. The ‘RegistrarName’ can be any string.
- Otherwise, the ‘servername’ is equal to ‘RegistrarIP’ (either FQDN or numerical IP address), if configured.
- Otherwise the ‘servername’ is equal to ‘ProxyName’ if configured. The ‘ProxyName’ can be any string.
- Otherwise the ‘servername’ is equal to ‘ProxyIP’ (either FQDN or numerical IP address).

The parameter ‘GWRegistrationName’ can be any string. If the parameter is not defined, the parameter ‘UserName’ is used instead.

The ‘sipgatewayname’ parameter (defined in the *ini* file or set from the Web browser), can be any string. Some Proxy servers require that the ‘sipgatewayname’ (in REGISTER messages) is set equal to the Registrar / Proxy IP address or to the Registrar / Proxy domain name.

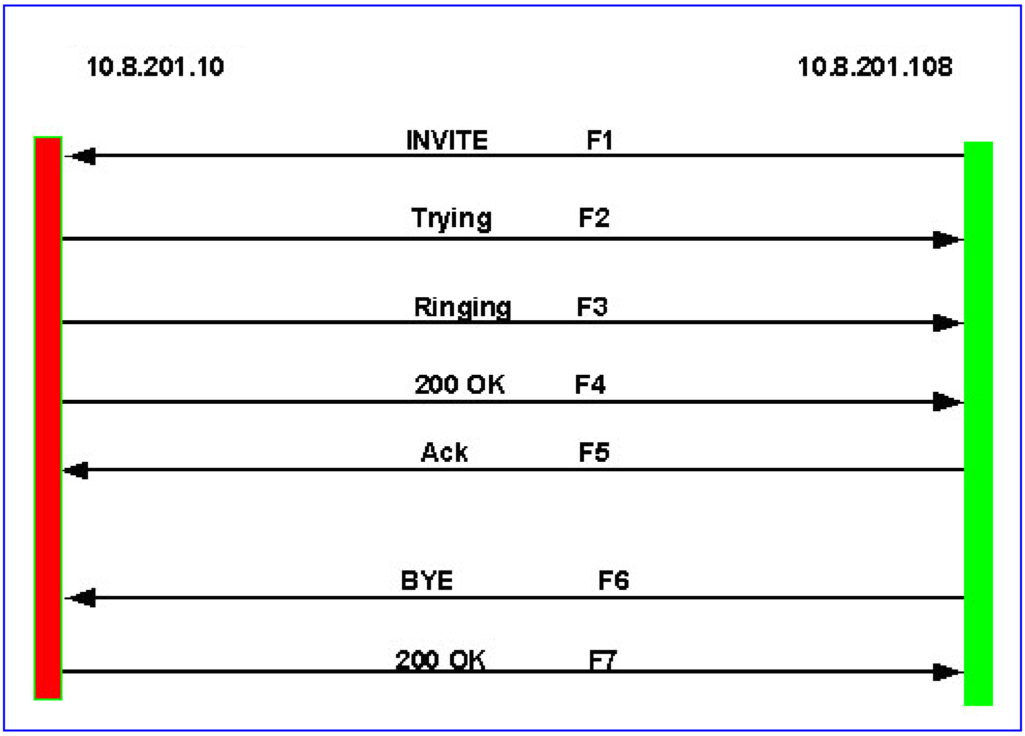
REGISTER messages are sent to the Registrar’s IP address (if configured) or to the Proxy’s IP address. A single message is sent once per gateway, or messages are sent per B-channel according to the parameter ‘AuthenticationMode’. The registration request is resent according to the parameter ‘RegistrartionTimeDivider’. For example, if ‘RegistrationTimeDivider = 70’ (%) and Registration Expires time = 3600, the gateway resends its registration request after 3600 x 70% = 2520 sec. The default value of ‘RegistrartionTimeDivider’ is 50%.

If registration per B-channel is selected, on gateway startup, the gateway sends REGISTER requests according to the maximum number of allowed SIP dialogs (configured by the parameter NumberOfActiveDialogs). After each received response, the subsequent REGISTER request is sent.

8.12 SIP Call Flow Example

The SIP call flow, shown in Figure 8-2, describes SIP messages exchanged between a Mediant 2000 gateway and an MP-108 gateway during a simple call. MP-108 (10.8.201.108) with phone number '8000', calls Mediant 2000 (10.8.201.10) with phone number '1000':

Figure 8-2: SIP Call Flow Example



F1 10.8.201.108 ==> 10.8.201.10 INVITE

```
INVITE sip:1000@10.8.201.10;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.8.201.108;branch=z9hG4bKacsiJkDGd
From: <sip:8000@10.8.201.108>;tag=1c5354
To: <sip:1000@10.8.201.10>
Call-ID: 534366556655skKw-8000--1000@10.8.201.108
CSeq: 18153 INVITE
Contact: <sip:8000@10.8.201.108;user=phone>
User-Agent: Audiocodes-Sip-Gateway/MP-108 FXS/v.4.20.299.410
Supported: 100rel,em
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO
Content-Type: application/sdp
Content-Length: 208

v=0
o=AudiocodesGW 18132 74003 IN IP4 10.8.201.108
s=Phone-Call
c=IN IP4 10.8.201.108
t=0 0
m=audio 4000 RTP/AVP 8 96
a=rtpmap:8 pcma/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F2 10.8.201.10 ==> 10.8.201.108 Trying

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.8.201.108;branch=z9hG4bKacsiJkDGd
From: <sip:8000@10.8.201.108>;tag=1c5354
To: <sip:1000@10.8.201.10>
Call-ID: 534366556655skKw-8000--1000@10.8.201.108
Server: Audiocodes-Sip-Gateway/TrunkPack 1610/v.4.20.299.412
CSeq: 18153 INVITE
Content-Length: 0
```

F3 10.8.201.10 ==> 10.8.201.108 180 Ringing

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.8.201.108;branch=z9hG4bKacsiJkDGd
From: <sip:8000@10.8.201.108>;tag=1c5354
To: <sip:1000@10.8.201.10>;tag=1c7345
Call-ID: 534366556655skKw-8000--1000@10.8.201.108
Server: Audiocodes-Sip-Gateway/TrunkPack 1610/v.4.20.299.412
CSeq: 18153 INVITE
Supported: 100rel,em
Content-Length: 0
```

Note: Phone ‘1000’ answers the call, and sends 200 OK message to MP gateway 10.8.201.108.

F4 10.8.201.10 ==> 10.8.201.108 200 OK

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.8.201.108;branch=z9hG4bKacsiJkDGd
From: <sip:8000@10.8.201.108>;tag=1c5354
To: <sip:1000@10.8.201.10>;tag=1c7345
Call-ID: 534366556655skKw-8000--1000@10.8.201.108
CSeq: 18153 INVITE
Contact: <sip:1000@10.8.201.10;user=phone>
Server: Audiocodes-Sip-Gateway/TrunkPack 1610/v.4.20.299.412
Supported: 100rel,em
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO
Content-Type: application/sdp
Content-Length: 206

v=0
o=AudiocodesGW 30221 87035 IN IP4 10.8.201.10
s=Phone-Call
c=IN IP4 10.8.201.10
t=0 0
m=audio 7210 RTP/AVP 8 96
a=rtpmap:8 pcma/8000
a=ptime:20
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
```

F5 10.8.201.108 ==> 10.8.201.10 ACK

```
ACK sip:1000@10.8.201.10;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.8.201.108;branch=z9hG4bKacZYpJWxZ
From: <sip:8000@10.8.201.108>;tag=1c5354
To: <sip:1000@10.8.201.10>;tag=1c7345
Call-ID: 534366556655skKw-8000--1000@10.8.201.108
User-Agent: Audiocodes-Sip-Gateway/MP-108 FXS/v.4.20.299.410
CSeq: 18153 ACK
Supported: 100rel,em
Content-Length: 0
```

Note: Phone ‘8000’ goes on-hook; gateway 10.8.201.108 sends BYE to gateway 10.8.201.10. Voice path is established.

F6 10.8.201.108 ==> 10.8.201.10 BYE

```
BYE sip:1000@10.8.201.10;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.8.201.108;branch=z9hG4bKacRKCVBud
From: <sip:8000@10.8.201.108>;tag=1c5354
To: <sip:1000@10.8.201.10>;tag=1c7345
Call-ID: 534366556655skKw-8000--1000@10.8.201.108
User-Agent: Audiocodes-Sip-Gateway/MP-108 FXS/v.4.20.299.410
CSeq: 18154 BYE
Supported: 100rel,em
Content-Length: 0F7 10.2.37.10 ==> 10.2.37.20     200 OK
```

F7 10.8.201.10 ==> 10.8.201.108 200 OK

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.8.201.108;branch=z9hG4bKacRKCVBud
From: <sip:8000@10.8.201.108>;tag=1c5354
To: <sip:1000@10.8.201.10>;tag=1c7345
Call-ID: 534366556655skKw-8000--1000@10.8.201.108
Server: Audiocodes-Sip-Gateway/TrunkPack 1610/v.4.20.299.412
CSeq: 18154 BYE
Supported: 100rel,em
Content-Length: 0
```

8.13 SIP Authentication Example

The gateway supports basic and digest authentication types, according to SIP RFC 3261 standard. A proxy server might require authentication before forwarding an INVITE message. A Registrar/Proxy server may also require authentication for client registration. A proxy replies to an unauthenticated INVITE with a 407 Proxy Authorization Required response, containing a Proxy-Authenticate header with the form of the challenge. After sending an ACK for the 407, the user agent can then resend the INVITE with a Proxy-Authorization header containing the credentials.

User agent, Redirect or Registrar servers typically use 401 Unauthorized responses to challenge authentication containing a WWW-Authenticate header, and expect the re-INVITE to contain an Authorization header.

The following example describes the Digest Authentication procedure including computation of user agent credentials.

The REGISTER request is sent to Registrar/Proxy server for registration, as follows:

```
REGISTER sip:10.2.2.222 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.200
From: <sip: 122@10.1.1.200>;tag=1c17940
To: <sip: 122@10.1.1.200>
Call-ID: 634293194@10.1.1.200
User-Agent: Audiocodes-Sip-Gateway/TrunkPack 1610/v.4.20.299.412
CSeq: 1 REGISTER
Contact: sip:122@10.1.1.200:
Expires:3600
```

On receiving this request the Registrar/Proxy returns 401 Unauthorized response.

```
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 10.2.1.200
From: <sip:122@10.2.2.222 >;tag=1c17940
To: <sip:122@10.2.2.222 >
Call-ID: 634293194@10.1.1.200
Cseq: 1 REGISTER
Date: Mon, 30 Jul 2001 15:33:54 GMT
Server: Columbia-SIP-Server/1.17
Content-Length: 0
WWW-Authenticate: Digest realm="audiocodes.com",
nonce="11432d6bce58ddf02e3b5e1c77c010d2",
stale=FALSE,
algorithm=MD5
```

According to the sub-header present in the WWW-Authenticate header the correct REGISTER request is formed.

Since the algorithm used is MD5, take:

The username from the ini file: M2K-AudioCodes

The realm return by the proxy: audiocodes.com

The password from the ini file: AudioCodes.

The equation to be evaluated: (according to RFC this part is called A1).

'M2K-AudioCodes:audiocodes.com:AudioCodes'.

The MD5 algorithm is run on this equation and stored for future usage.

The result is: ‘a8f17d4b41ab8dab6c95d3c14e34a9e1’

Next we need to evaluate the par called A2. We take:

The method type ‘REGISTER’

Using SIP protocol ‘sip’

Proxy IP from ini file ‘10.2.2.222’

The equation to be evaluated:

‘REGISTER:sip:10.2.2.222’.

The MD5 algorithm is run on this equation and stored for future usage.

The result is:’a9a031cfddcb10d91c8e7b4926086f7e’

The final stage:

The A1 result

The nonce from the proxy response: ‘11432d6bce58ddf02e3b5e1c77c010d2’

The A2 result

The equation to be evaluated:

‘A1:11432d6bce58ddf02e3b5e1c77c010d2:A2’.

The MD5 algorithm is run on this equation. The outcome of the calculation is the response needed by the gateway to be able to register with the Proxy.

The response is: ‘b9c45d0234a5abf5ddf5c704029b38cf’

At this time a new REGISTER request is issued with the response:

```
REGISTER sip:10.2.2.222 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.200
From: <sip: 122@10.1.1.200>;tag=1c23940
To: <sip: 122@10.1.1.200>
Call-ID: 654982194@10.1.1.200
Server: Audiocodes-Sip-Gateway/TrunkPack 1610/v.4.20.299.412
CSeq: 1 REGISTER
Contact: sip:122@10.1.1.200:
Expires:3600
Authorization: Digest, Username: MP108-AudioCodes,
realm="audiocodes.com”,
nonce="11432d6bce58ddf02e3b5e1c77c010d2",
uri=”10.2.2.222”,
response=“ b9c45d0234a5abf5ddf5c704029b38cf”
```

On receiving this request, if accepted by the Proxy, the proxy returns a 200 OK response closing the REGISTER transaction.

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.1.1.200
From: <sip: 122@10.1.1.200>;tag=1c23940
To: <sip: 122@10.1.1.200>
Call-ID: 654982194@10.1.1.200
Cseq: 1 REGISTER
Date: Thu, 26 Jul 2001 09:34:42 GMT
Server: Columbia-SIP-Server/1.17
Content-Length: 0
Contact: <sip:122@10.1.1.200>; expires="Thu, 26 Jul 2001 10:34:42 GMT";
action=proxy; q=1.00
Contact: <122@10.1.1.200:>; expires="Tue, 19 Jan 2038 03:14:07 GMT";
action=proxy; q=0.00
Expires: Thu, 26 Jul 2001 10:34:42 GMT
```

Reader's Notes

9 Networking Capabilities

9.1 Ethernet Interface Configuration

Using the parameter ‘EthernetPhyConfiguration‘, users can control the Ethernet connection mode.

Either the manual modes (10 Base-T Half-Duplex, 10 Base-T Full-Duplex, 100 Base-TX Half-Duplex, 100 Base-TX Full-Duplex) or Auto-Negotiate mode can be used.

Auto-Negotiation falls back to Half-Duplex mode when the opposite port is not Auto-Negotiate, but the speed (10 Base-T, 100 Base-TX) in this mode is always configured correctly. Note that configuring the gateway to Auto-Negotiate mode while the opposite port is set manually to Full-Duplex (either 10 Base-T or 100 Base-TX) is invalid (as it causes the gateway to fall back to Half-Duplex mode while the opposite port is Full-Duplex). It is also invalid to set the gateway to one of the manual modes while the opposite port is either Auto-Negotiate or not exactly matching (both in speed and in duplex mode). Users are encouraged to always prefer Full-Duplex connections to Half-Duplex ones and 100 Base-TX to 10 Base-T (due to the larger bandwidth). It is strongly recommended to use the same mode in both link partners. Any mismatch configuration can yield unexpected functioning of the Ethernet connection.

Note that when remote configuration is performed, the gateway should be in the correct Ethernet setting prior to the time this parameter takes effect. When, for example, the gateway is configured using BootP/TFTP, the gateway must perform many Ethernet-based transactions prior to reading the *ini* file containing this gateway configuration parameter.

To work around this problem, the gateway always uses the last Ethernet setup mode configured. This way, if users want to configure the gateway to work in a new network environment in which the current Ethernet setting of the gateway is invalid, they should first modify this parameter in the current network so that the new setting holds next time gateway is restarted. After reconfiguration has completed, connect the gateway to the new network and restart it. As a result, the remote configuration process that takes place in the new network uses a valid Ethernet configuration.

9.2 Ethernet Interface Redundancy

The Mediant 2000 supports the following redundancy scheme:

At the beginning of the start-up procedure, the gateway tests whether the ‘Primary’ Ethernet interface is connected by checking the existence of the Ethernet link carrier. If it is connected, the start-up procedure commences as usual. If not, the start-up application tries the ‘Secondary’ Ethernet interface. If this interface is connected, the whole start-up procedure is performed using it. If both interfaces are out of order, the start-up procedure commences using the parameters, tables, and software residing in the gateway’s non-volatile memory. Note that Ethernet switchover occurs only once during the start-up procedure (at its beginning). If the Ethernet interface fails after the selection is made, the gateway does not switch over to the second port.

After start-up has completed and the operational software is running, the gateway continues to use the Ethernet port used for program load. The gateway switches over from one Ethernet port to the other one every time an Ethernet link carrier loss is detected on the active Ethernet port, and if the Ethernet link of the other port is operational. Switchover takes place only once per link loss (that is, the ‘secondary’ interface stays the active one even if the ‘primary’ interface has returned to life).

After start-up, the gateway generates a gratuitous ARP message each time a switchover occurs.

For correct functionality of the redundancy mechanism, it is recommended to configure both links to the same mode. It is essential that both link partners (the primary link partner and the secondary link partner) have the same capabilities, so that whenever a switchover occurs the gateway is able to provide at least the same Ethernet services as were provided prior to the switchover.

For correct functionality of the redundancy mechanism, it is recommended to set the physical secondary link prior to the gateway being reset (since the MAC configuration cannot be changed thereafter).

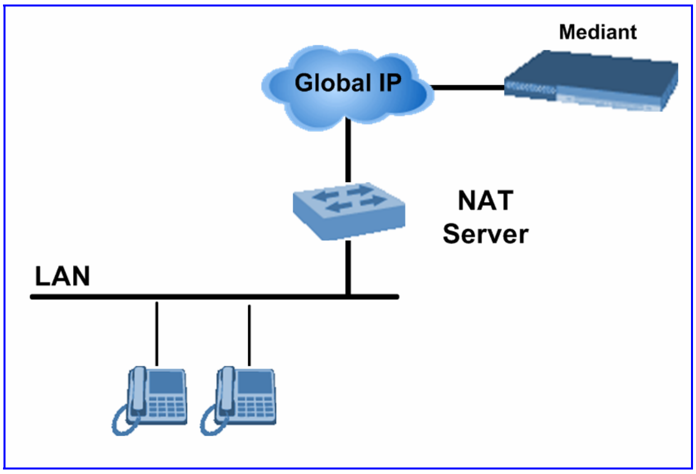
Note that as the two Ethernet ports use the same MAC address, the external switches connected to the gateways can in some cases create a noticeable switchover delay due to their internal switching logic, though on the gateway level, the switchover delay is minimal (milliseconds).

9.3 NAT Support

Network Address Translation (NAT) is a mechanism that maps a set of internal IP addresses used within a private network to global IP addresses, providing transparent routing to end hosts. The primary advantages of NAT are (1) reduces the number of global IP addresses required in a private network (global IP addresses are only used to connect to the Internet); (2) provides a better network security by hiding its internal architecture.

Figure 9-1 below illustrates the supported NAT architecture.

Figure 9-1: NAT Functioning



The way SIP is designed creates a problem for VoIP traffic to pass through NAT. SIP uses IP addresses and port numbers in its message body. The NAT server can't modify SIP messages and therefore, can't change local to global addresses.

Two different streams traverse through NAT: signaling and media. A gateway (located behind a NAT) that initiates a signaling path will have problems in receiving incoming signaling responses (they will be blocked by the NAT). Furthermore, the initiating gateway must notify the receiving gateway where to send the media to.

To solve these problems the following mechanisms are available:

- STUN (refer to Section 9.3.1 below).
- First Incoming Packet Mechanism (refer to Section 9.3.2 on page 232)
- RTP No-Op packets according to the avt-rtp-noop draft (refer to Section 9.3.3on page 232).
- For SNMP NAT traversal, refer to Section 15.10 on page 326.

9.3.1 STUN

Simple Traversal of UDP through NATs (STUN) (according to RFC 3489) is a client / server protocol that solves most of the NAT traversal problems. The STUN server operates in the public Internet and the STUN clients are embedded in end-devices (located behind NAT). STUN is used both for the signaling and the media streams. STUN works with many existing NAT types, and does not require any special behavior from them.

STUN enables the gateway to discover the presence (and types) of NATs and firewalls located between it and the public Internet. It provides the gateway with the capability to determine the public IP address and port allocated to it by the NAT. This information is later embedded in outgoing SIP/SDP messages and enables remote SIP user agents to reach the gateway. It also discovers the binding lifetime of the NAT (the refresh rate necessary to keep NAT 'Pinholes' open).

On startup the gateway sends a STUN Binding Request. The information received in the STUN Binding Response (IP address:port) is used for SIP signaling. This information is updated every NATBindingDefaultTimeout.

At the beginning of each call, if STUN is needed (i.e., not an internal NAT call), the media ports of the call are mapped. The call is delayed until the STUN Binding Response (that includes a global IP:port) for each media (RTP, RTCP and T.38) is received.

To enable STUN:

- Set the parameter EnableSTUN to 1
- Define the STUN server address using one of the following methods:
    - Define the IP address of the primary and the secondary (optional) STUN servers using the parameters STUNServerPrimaryIP and STUNServerSecondaryIP. If the primary STUN server isn't available, the gateway tries to communicate with the secondary server.
    - Define the domain name of the STUN server using the StunServerDomainName parameter. The STUN client retrieves all STUN servers with an SRV query to resolve this domain name to an IP address and port, sort the server list, and use the servers according to the sorted list.
- Use the parameter NATBindingDefaultTimeout to determine the default NAT binding lifetime in seconds. STUN is used to refresh the binding information after this time expires.

Notes:

- STUN only applies to UDP (doesn't support TCP and TLS).
- STUN can't be used when the gateway is located behind a symmetric NAT.
- For defining the STUN server, use either the STUNServerPrimaryIP or STUNServerDomainName parameter, with priority to the first one.

9.3.2 First Incoming Packet Mechanism

If the remote gateway resides behind a NAT device, it's possible that the gateway can activate the RTP/RTCP/T.38 streams to an invalid IP address / UDP port. To avoid such cases, the gateway automatically compares the source address of the incoming RTP/RTCP/T.38 stream with the IP address and UDP port of the remote gateway. If the two are not identical, the transmitter modifies the sending address to correspond with the address of the incoming stream. The RTP, RTCP and T.38 can thus have independent destination IP addresses and UDP ports.

Users can choose to disable the NAT mechanism by setting the *ini* file parameter 'DisableNAT' to 1. The two parameters 'EnableIpAddrTranslation' and 'EnableUdpPortTranslation' enable users to specify the type of compare operation that takes place on the first incoming packet. To compare only the IP address, set 'EnableIpAddrTranslation = 1' and 'EnableUdpPortTranslation = 0'. In this case, if the first incoming packet arrives with only a difference in the UDP port, the sending addresses won't change. If both the IP address and UDP port need to be compared, then both parameters need to be set to 1.

9.3.3 No-Op Packets

The gateway's No-Op packet support can be used to verify Real-Time Transport Protocol (RTP) and T.38 connectivity, and to keep NAT bindings and Firewall pinholes open. No-Op packets are available for sending in RTP and T.38 formats.

Users can control the activation of No-Op packets by using the *ini* file parameter NoOperationSendingMode. If No-Op packet transmission is activated, users can control the time interval in which No-Op packets are sent in the case of silence (i.e., no RTP or T.38 traffic). This is performed using the NoOpInterval *ini* parameter.

Note: Receipt of No-Op is always supported.

- **RTP No-Op:**

  The RTP No-Op support complies with IETF's draft-wing-avt-rtp-noop-03.txt (titled 'A No-Op Payload Format for RTP'). This IETF document defines a No-Op payload format for RTP.

  The draft defines the RTP payload type as dynamic. Users can control the payload type with which the No-Op packets are sent. This is performed using the RTPNoOpPayloadType *ini* parameter. AudioCodes' default payload type is 120.

- **T.38 No-Op:**

  T.38 No-Op packets are sent only while a T.38 session is activated. Sent packets are a duplication of the previously sent frame (including duplication of the sequence number).

9.4 Point-to-Point Protocol over Ethernet (PPPoE)

PPPoE is a method of sending the Point-to-Point Protocol over Ethernet network.

9.4.1 Point-to-Point Protocol (PPP) Overview

Point-to-Point Protocol (PPP) provides a method of transmitting data over serial point-to-point links. The protocol defines establishing, configuring and testing the data link connection and the network protocol.

The PPP standard describes a state machine used to establish a valid connection between two hosts over a serial connection. There are three major stages described, helping to establish a network layer (such as an IP) connection over the point-to-point link: LCP (Link Configuration Protocol) Authentication and NCP (Network Control Protocol). Once the network protocol is configured, the two hosts can communicate, sending network layer protocol (such as IP) over the PPP connection (a small PPP header is added at the beginning of each packet).

At the initial phase, the hosts use LCP (link configuration protocol) to negotiate for link characteristic and parameters. Packets sent in this phase have two octets of ‘PPP header’ followed by LCP message with variable length. Various parameters and options are negotiable at this phase, including MRU (maximum receive unit), Authentication Protocol, and others.

Once the link is established (each side sends a ‘configure ack’ message to the other side), the authentication phase may begin. The authentication phase is not mandatory. However, it is negotiated in the link configuration phase. A host may ask other hosts for authentication using Password Authentication Protocol (PAP) or Challenge Handshake Authentication Protocol (CHAP).

The PAP sends the username and password to the remote host unencrypted.

The CHAP is a more sophisticated method of authentication. The two hosts share a ‘secret’. The authenticator sends a ‘challenge’ to the host requesting authentication. The host performs a calculation (one-way hash) using the challenge received from the authenticator and the shared ‘secret’, and sends the result to the authenticator. The authenticator verifies the host if the result of the calculation is correct; otherwise it is rejected.

The last configuration phase, immediately after the authentication phase (or after the Link Configuration) is the Network Control Protocol. There is a family of control protocols for establishing and configuring different network-layer protocols, for example, IPCP (PPP Internet Protocol Control Protocol), IPv6CP (PPP IP v6 Control Protocol), and BCP (PPP Bridging Control Protocol). Each of them handles and manages the specific needs required by their respective network-layer protocol.

When working in an IP network, IPCP is used as the Network Configuration Protocol. The IPCP is used to configure the network layer of the hosts, requesting/declaring on IP Addresses.

Further information on PPP Protocol is available on the IETF website (http://www.ietf.org/rfc/rfc1661.txt). Further information on Password Authentication Protocol is available on the IETF website (http://www.ietf.org/rfc/rfc1334.txt). Further information on Challenge Handshake Authentication Protocol is available on the IETF website (http://www.ietf.org/rfc/rfc1994.txt). Further information on PPP Internet Protocol Control Protocol (IPCP) is available on the IETF website (http://www.ietf.org/rfc/rfc1332.txt).

9.4.2 PPPoE Overview

PPPoE is a method of sending the Point-to-Point Protocol over Ethernet network. PPPoE provides the ability to connect a network of hosts over a simple bridging access device to a remote Access Concentrator. Access control, billing and type of service can be done on a per-user, rather than a per-site, basis.

A common use of the PPPoE is in the ADSL market: The home PC is connected to a modem via Ethernet, and the PC uses the PPPoE to 'simulate' as if it was directly connected to the remote host on a point-to-point connection.

Since PPPoE frames are sent over Ethernet, each PPP session must learn the Ethernet address of the remote peer, as well as establish a unique session identifier. The PPPoE standard describes a discovery protocol that provides this. A PPPoE session begins with a discovery phase. Only after this discovery is completed can the PPP state machine start (with LCP, Authentication etc, as described above).

Each of the Ethernet frames carrying PPP session has a standard Ethernet header followed by PPPoE header, and is sent with the remote host Ethernet MAC address (except for the very first one, in the discovery phase, which is broadcasted to all hosts).

Further information on the transmission of PPPoE is available on the IETF website (http://www.ietf.org/rfc/rfc2516.txt).

9.4.3 PPPoE in AudioCodes Gateways

The AudioCodes gateway contains a PPPoE client embedded in its software. When correctly configured (see *ini* file parameters) the gateway can try to connect to a remote PPPoE Access Concentrator.

When restarting the gateway after several BOOTP attempts, if PPPoE is enabled (see *ini* file parameter EnablePPPoE), the gateway tries to initiate a PPP session.

The gateway initiates a PPPoE discovery phase to discover a PPPoE Access Concentrator. It does this by broadcasting a discovery initialization packet (PADI). If an Access Concentrator exists and replies, the gateway tries to connect to this Access Concentrator. If this initial connection succeeds, then the PPP LCP phase starts - each side of the PPPoE connection sends LCP configuration requests to configure the PPP link.

The gateway PPPoE client supports both PAP and CHAP authentications. The type of authentication protocol used is according to the request from the authentication server. In the LCP configuration phase, the server requires a specific authentication (none, PAP, or CHAP are supported). The *ini* file parameters PPPoEUserName, PPPoEPassword, and PPPoEServerName are used to configure the authentication parameters. If the Access Concentrator is configured to operate in PAP, the PPPoEUserName and PPPoEPassword are used as Username and Password (in this case, the PPPoEServerName parameter is not used). If the Access Concentrator is configured to operate in CHAP, the PPPoEUserName parameter functions as Client Name (sent in the CHAP response packet), while the PPPoEPassword functions as the shared secret (calculated along with the challenge to produce the response). In this case, the PPPoEServerName is the name of the server. Some hosts can be configured to authenticate to multiple servers. In such hosts, the server name is used to identify the "secret" that should be used.

Note: The AudioCodes gateway, being a PPPoE client, requests no authentication.

After the gateway has been authenticated, it needs to configure a network layer protocol. The gateway uses the IP protocol. Therefore, the used NCP will be IPCP (IP Configuration Protocol). In this phase, if the *ini* file parameter PPPoEStaticIPAddress is defined, the gateway requests the remote host to assign this address for its use.

When working in a PPPoE environment, the gateway negotiates for its IP address (as described above). However, if the user desires to disable the PPPoE client, the gateway can be configured to use default values for IP address, subnet mask and default gateway. This can be done using *ini* file parameters PPPoERecoverIPAddress, PPPoERecoverSubnetMask and PPPoERecoverDfgwAddress. These parameters indicate to the gateway that if the PPPoE is disabled and no BootP server is activated, as required in the gateway to use a PPPoE environment, then the gateway should use these defaults for its IP configuration.

For a detailed description of the *ini* file parameters for PPPoE, refer to Section 6.6 on page 130.

Note: When working with a PPPoE server (Access Concentrator) that does not reply to LCP Echo messages (which by default, the gateway periodically sends) you may want to disable the LCP Echo messages by using the *ini* file parameter PPPoELCPEchoEnable. (For a description of this parameter, refer to Section 6.6 on page 130.)

9.5 IP Multicasting

The gateway supports IP Multicasting level 1 according to RFC 2236 (i.e. IGMP version 2) for RTP channels. The gateway is capable of transmitting and receiving Multicast packets.

9.6 Robust Reception of RTP Streams

This mechanism filters out unwanted RTP streams that are sent to the same port number on the gateway. These multiple RTP streams can result from traces of previous calls, call control errors and deliberate attacks.

When more than one RTP stream reaches the gateway on the same port number, the gateway accepts only one of the RTP streams and rejects the rest of the streams. The RTP stream is selected according to the following procedure:

The first packet arriving on a newly opened channel sets the source IP address and UDP port from which further packets are received. Thus, the source IP address and UDP port identify the currently accepted stream. If a new packet arrives whose source IP address or UDP port are different to the currently accepted RTP stream, there are two options:

- The new packet has a source IP address and UDP port which are the same as the remote IP address and UDP port that were stated during the opening of the channel. In this case, the gateway reverts to this new RTP stream.
- The new packet has any other source IP address and UDP port, in which case the packet is dropped.

9.7 Multiple Routers Support

Multiple routers support is designed to assist the media gateway when it operates in a multiple routers network. The gateway learns the network topology by responding to ICMP redirections and caches them as routing rules (with expiration time).

When a set of routers operating within the same subnet serve as gateways to that network and intercommunicate using a dynamic routing protocol, the routers can determine the shortest path to a certain destination and signal the remote host the existence of the better route. Using multiple router support the media gateway can utilize these router messages to change its next hop and establish the best path.

Note: Multiple Routers support is an integral feature that doesn’t require configuration.

9.8 Simple Network Time Protocol Support

The Simple Network Time Protocol (SNTP) client functionality generates requests and reacts to the resulting responses using the NTP version 3 protocol definitions (according to RFC 1305). Through these requests and responses, the NTP client is able to synchronize the system time to a time source within the network, thereby eliminating any potential issues should the local system clock 'drift' during operation. By synchronizing time to a network time source, traffic handling, maintenance, and debugging actions become simplified for the network administrator.

The NTP client follows a simple process in managing system time; the NTP client requests an NTP update, receives an NTP response, and updates the local system clock based on a configured NTP server within the network.

The client requests a time update from a specified NTP server at a specified update interval. In most situations this update interval should be every 24 hours based on when the system was restarted. The NTP server identity (as an IP address) and the update interval are configurable parameters that can be specified either in the *ini* file (NTPServerIP, NTPUpdateInterval respectively) or via an SNMP MIB object.

When the client receives a response to its request from the identified NTP server it must be interpreted based on time zone, or location, offset that the system is to a standard point of reference called the Universal Time Coordinate (UTC). The time offset that the NTP client should use is a configurable parameter that can be specified either in the *ini* file (NTPServerUTCOffset) or via an SNMP MIB object.

If required, the clock update is performed by the client as the final step of the update process. The update is done in such a way as to be transparent to the end users. For instance, the response of the server may indicate that the clock is running too fast on the client. The client slowly robs bits from the clock counter in order to update the clock to the correct time. If the clock is running too slow, then in an effort to catch the clock up, bits are added to the counter, causing the clock to update quicker and catch up to the correct time. The advantage of this method is that it does not introduce any disparity in the system time, that is noticeable to an end user, or that could corrupt call timeouts and timestamps.

9.9 IP QoS via Differentiated Services (DiffServ)

DiffServ is architecture providing different types or levels of service for IP traffic. DiffServ (according to RFC 2474) offers the capability to prioritize certain traffic types, depending on their priority, thereby accomplishing a higher-level QoS at the expense of other traffic types. By prioritizing packets, DiffServ routers can minimize transmission delays for time-sensitive packets such as VoIP packets.

The gateway can be configured to set a different DiffServ value to IP packets according to their class-of-service (Network, Premium Media, Premium Control, Gold and Bronze).

For the mapping of an application to its class-of-service, refer to Table 9-1 on page 238.

The DiffServ parameters are described in Table 6-1 on page 130.

9.10 VLANS and Multiple IPs

9.10.1 Multiple IPs

Media, Control, and Management (OAM) traffic in the gateway can be assigned one of the following IP addressing schemes:

- Single IP address for all traffic (i.e., Media, Control, and OAM).
- Separate IP address for each traffic type.

  For separate IP addresses, the different traffic types are separated into three dedicated networks. Instead of a single IP address, the gateway is assigned three IP addresses and subnet masks, each relating to a different traffic type. This architecture enables users to integrate the gateway into a three-network environment that is focused on security and segregation. Each entity in the gateway (e.g., Web and RTP) is mapped to a single traffic type (according to Table 9-1 on page 238) in which it operates.
- Two separate IP addresses (Dual IP mode)--one for a specific traffic type and the other for a combination of two traffic types.

  In Dual IP mode, the gateway is assigned two IP addresses for the different traffic types. One IP address is assigned to a combination of two traffic types (Media and Control, OAM and Control, or OAM and Media), while the other IP address is assigned to whichever traffic type that is not included in this combination. For example, a typical scenario using this mode would include one IP address assigned for Control and OAM, and another IP address assigned for Media.

Notes:

- A default gateway is supported only for the Media traffic type; for the other two, use the IP Routing table.
- The IP address and subnet mask used in the Single IP Network mode are carried over to the OAM traffic type in the Multiple IP Network mode.

For detailed information on integrating the gateway into a VLAN and multiple IPs network, refer to Section 9.10.3 on page 239. For detailed information on configuring the multiple IP parameters, refer to Table 6-1 on page 130.

9.10.2 IEEE 802.1p/Q (VLANs and Priority)

The Virtual Local Area Network (VLAN) mechanism enables the gateway to be integrated into a VLAN-aware environment that includes switches, routers and endpoints.

When in VLAN-enabled mode, each packet is tagged with values that specify its priority (class-of-service) (IEEE 802.1p) and the identifier (traffic type) of the VLAN to which it belongs (media, control or management) (IEEE 802.1Q).

The class-of-service mechanism can be utilized to accomplish Ethernet QoS. Packets sent by the gateway to the Ethernet network are divided into five, different-priority classes (Network, Premium media, Premium control, Gold and Bronze). The priority of each class is determined by a corresponding *ini* file parameter.

Traffic type tagging can be used to implement Layer 2 VLAN security. By discriminating traffic into separate and independent domains, the information is preserved within the VLAN. Incoming packets received from an incorrect VLAN are discarded.

For the mapping of an application to its class-of-service and traffic type, refer to Table 9-1 below.

Media traffic type is assigned 'Premium media' class of service, Management traffic type is assigned 'Bronze' class of service, and Control traffic type is assigned 'Premium control' class of service. For example, RTP/RTCP traffic is assigned the Media VLAN ID and 'Premium media' class of service, whereas Web traffic is assigned the Management VLAN ID and 'Bronze' class of service. Each of these parameters can be configured with an 802.1p/q value: traffic type to VLAN ID, and class of service to 802.1p priority.

Notes:

- As a safety measure, the VLAN mechanism is activated only when the gateway is loaded from the flash memory. Therefore, when using BootP: Load an *ini* file with 'VlanMode = 1' and 'SaveConfiguration = 1'. Then (after the gateway is active) reset the gateway with TFTP disabled, or by using any method except for BootP.
- The gateway must be connected to a VLAN-aware switch, and the switch's PVID must be equal to the gateway's native VLAN ID.

For information on how to configure VLAN parameters, refer to Table 6-1 on page 130.

Table 9-1: Traffic / Network Types and Priority

Application	Traffic / Network Types	Class-of-Service (Priority)
Debugging interface	Management	Bronze
Telnet	Management	Bronze
DHCP	Management	Network
Web server (HTTP)	Management	Bronze
SNMP GET/SET	Management	Bronze
Web server (HTTPS)	Management	Bronze
IPSec IKE	Determined by the service	Determined by the service
RTP traffic	Media	Premium media
RTCP traffic	Media	Premium media
T.38 traffic	Media	Premium media
SIP	Control	Premium control
SIP over TLS (SIPS)	Control	Premium control
Syslog	Management	Bronze
ICMP	Management	Determined by the initiator of the request
ARP listener	Determined by the initiator of the request	Network
SNMP Traps	Management	Bronze
DNS client	EnableDNSasOAM	Network
NTP	EnableNTPasOAM	Depends on the traffic type: Control: Premium control Management: Bronze
NFS	NFSServers_VlanType in the NFSServers table	Gold

9.10.2.1 Operation

- **Outgoing packets (from the gateway to the switch):**

  All outgoing packets are tagged, each according to its interface (control, media or OAM). If the gateway's native ID is identical to one of the other IDs (usually to the OAM ID), this ID (e.g., OAM) is set to zero on outgoing packets (VlanSendNonTaggedOnNative = 0). This method is called Priority Tagging (p tag without Q tag). If the parameter VlanSendNonTaggedOnNative is set to 1, the gateway sends regular packets (with no VLAN tag).

- **Incoming packets (from the switch to the gateway):**

  The switch sends all packets intended for the gateway (according to the switch's configuration) to the gateway without altering them. For packets whose VLAN ID is identical to the switch's PVID. In this case, the switch removes the tag and sends a packet.

  The gateway only accepts packets that have a VLAN ID identical to one of its interfaces (control, media or OAM). Packets with a VLAN ID that is 0 or packets without a tag are accepted only if the gateway's native VLAN ID is identical to the VLAN ID of one of its interfaces. In this case, the packets are sent to the relevant interface. All other packets are rejected.

9.10.3 Getting Started with VLANS and Multiple IPs

By default, the gateway operates without VLANs and multiple IPs, using a single IP address, subnet mask and default gateway IP address. This section provides an example of the configuration required to integrate the gateway into a VLAN and multiple IPs network using the Embedded Web Server (refer to Section 9.10.3.1) and *ini* file (refer to Section 9.10.3.2 on page 242). Table 9-2 below shows an example configuration that is implemented in the following sections.

Table 9-2: Example of VLAN and Multiple IPs Configuration

Network Type	IP Address	Subnet Mask	Default Gateway IP Address	VLAN ID	External Routing Rule
OAM	10.31.174.50	255.255.0.0	0.0.0.0	4	83.4.87.X
Control	10.32.174.50	255.255.0.0	0.0.0.0	5	130.33.4.6
Media	10.33.174.50	255.255.0.0	10.33.0.1	6	--

Note that since a default gateway is available only for the Media network, for the gateway to be able to communicate with an external device / network on its OAM and Control networks, IP routing rules must be used.

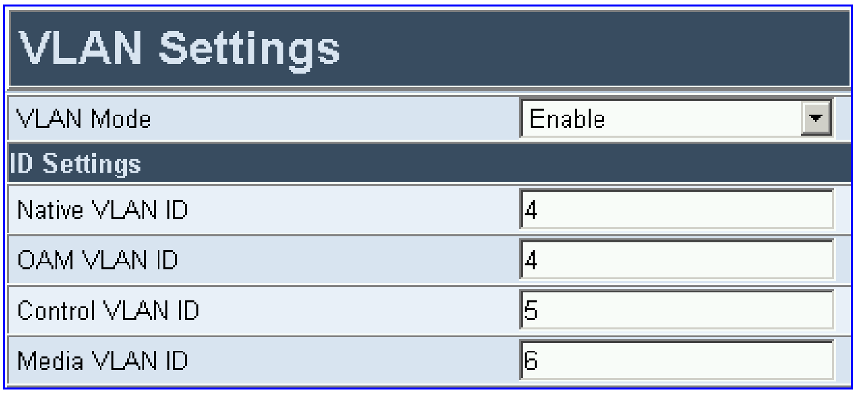
Note: The values provided in Sections 9.10.3.1 and 9.10.3.2 are sample parameter values only and are to be replaced with actual values appropriate to your system.

9.10.3.1 Integrating Using the Embedded Web Server

➢ **To integrate the gateway into a VLAN and multiple IPs network using the Embedded Web Server, take these 7 steps:**

1. Access the Embedded Web Server (Section 5.3 on page 58).
2. Use the Software Upgrade Wizard (Section 5.8.1 on page 115) to load and *burn* the firmware version to the gateway (VLANs and multiple IPs support is available only when the firmware is burned to flash).
3. Configure the VLAN parameters by completing the following steps:
   - Open the ‘VLAN Settings’ screen (**Advanced Configuration** menu > **Network Settings** > **VLAN Settings** option); the ‘VLAN Settings’ screen is displayed.
   - Modify the VLAN parameters to correspond to the values shown in Figure 9-2 below.

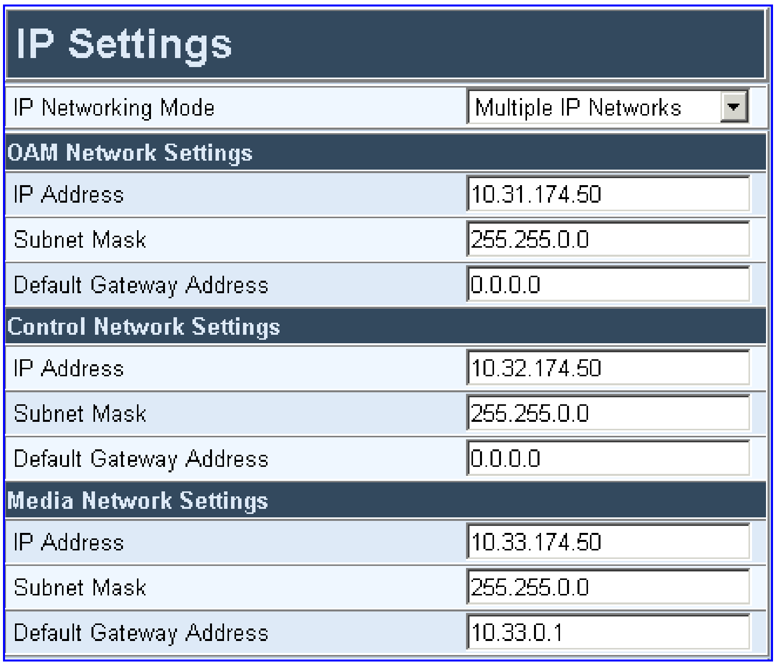
Figure 9-2: Example of the VLAN Settings Screen



   - Click the **Submit** button to save your changes.
4. Configure the multiple IP parameters by completing the following steps:
   - Open the ‘IP Settings’ screen (**Advanced Configuration** menu > **Network Settings** > **IP Settings** option); the ‘IP Settings’ screen is displayed.
   - Modify the IP parameters to correspond to the values shown in Figure 9-3. Note that the OAM, Control and Media Network Settings parameters appear only after you select the options ‘Multiple IP Networks’ or 'Dual IP' in the field ‘IP Networking Mode’.

> **Note:** Configure the OAM parameters only if the OAM networking parameters are different from the networking parameters used in the Single IP Network mode.

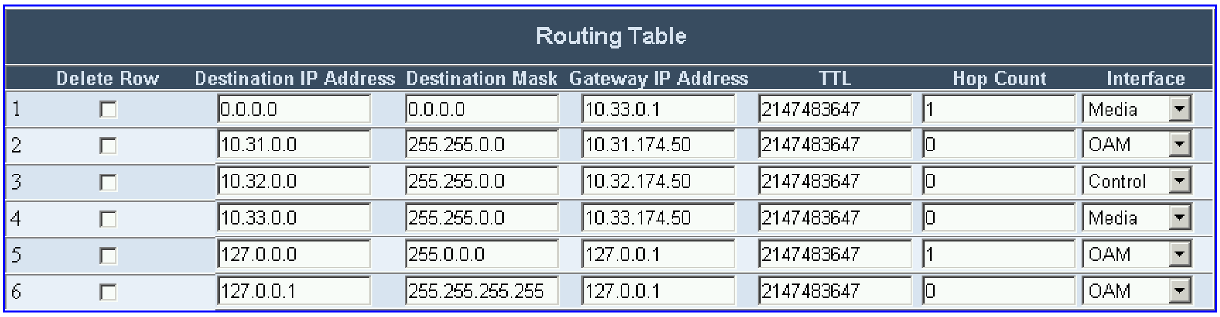
Figure 9-3: Example of the IP Settings Screen



- Click the **Submit** button to save your changes.

5. Configure the IP Routing table by completing the following steps (the IP Routing table is required to define static routing rules for the OAM and Control networks since a default gateway isn't supported for these networks):
   - Open the 'IP Routing Table' screen (**Advanced Configuration** menu > **Network Settings** > **IP Routing Table** option); the 'IP Routing Table' screen is displayed.

Figure 9-4: Example of the IP Routing Table Screen



   - Use the 'Add a new table entry' pane to add the routing rules shown in Table 9-3 below.

Table 9-3: Example of IP Routing Table Configuration

Destination IP Address	Destination Mask	Gateway IP Address	Hop Count	Network Type
130.33.4.6	255.255.255.255	10.32.0.1	20	Control
83.4.87.6	255.255.255.0	10.31.0.1	20	OAM

   - Click the **Submit** button to save your changes.

6. Save your changes to flash so they are available after a power fail, refer to Section 5.9.2 on page 124.
7. Reset the gateway (refer to Section 5.9.3 on page 125).

9.10.3.2 Integrating Using the *ini* File

➢ **To integrate the gateway into a VLAN and multiple IPs network using the *ini* file, take these 3 steps:**

1. Prepare an *ini* file with parameters shown in Figure 6-1 (refer to the following notes):
   - If the BootP/TFTP utility and the OAM interface are located in the same network, the Native VLAN ID (VlanNativeVlanId) must be equal to the OAM VLAN ID (VlanOamVlanId), which in turn must be equal to the PVID of the switch port the gateway is connected to. Therefore, set the PVID of the switch port to 4 (in this example).
   - Configure the OAM parameters (LocalOAMPAddress, LocalOAMSubnetMask and LocalOAMDefaultGW) only if the OAM networking parameters are different from the networking parameters used in the Single IP Network mode.
   - The IP Routing table is required to define static routing rules for the OAM and Control networks since a default gateway isn't supported for these networks.

Figure 9-5: Example of VLAN and Multiple IPs *ini* File Parameters

```
; VLAN Configuration
VlanMode=1
VlanOamVlanId=4
VlanNativeVlanId=4
VlanControlVlanId=5
VlanMediaVlanID=6

; Multiple IPs Configuration
EnableMultipleIPs=1
LocalMediaIPAddress=10.33.174.50
LocalMediaSubnetMask=255.255.0.0
LocalMediaDefaultGW=10.33.0.1
LocalControlIPAddress=10.32.174.50
LocalControlSubnetMask=255.255.0.0
LocalControlDefaultGW=0.0.0.0
LocalOAMPAddress=10.31.174.50
LocalOAMSubnetMask=255.255.0.0
LocalOAMDefaultGW=0.0.0.0

; IP Routing table parameters
RoutingTableDestinationsColumn = 130.33.4.6, 83.4.87.6
RoutingTableDestinationMasksColumn = 255.255.255.255 , 255.255.255.0
RoutingTableGatewaysColumn = 10.32.0.1 , 10.31.0.1
RoutingTableInterfacesColumn = 1 , 0
RoutingTableHopsCountColumn = 20,20
```

2. Use the BootP/TFTP utility (Section D.6 on page 354) to load and *burn* (-fb option) the firmware version and the *ini* file you prepared in the previous step to the gateway (VLANs and multiple IPs support is available only when the firmware is burned to flash).
3. Reset the gateway after disabling it on the BootP/TFTP utility.

10 Advanced PSTN Configuration

10.1 Gateway Clock Settings

The gateway can either generate its own timing signals, using an internal clock, or recover them from one of the E1/T1 trunks.

- To use the internal gateway clock source configure the following parameters:
  - TDMBusClockSource = 1
  - ClockMaster = 1 (for all gateway trunks)
- To use the recovered clock option, configure the following parameters:
  - TDMBusClockSource = 4
  - ClockMaster_x = 0 (for all ‘slave’ gateway trunks connected to PBX#1)
  - ClockMaster_x = 1 (for all ‘master’ gateway trunks connected to PBX#2)

Assuming that the gateway recovers its internal clock from one of the ‘slave’ trunks connected to PBX#1, and provides clock to PBX#2 on its ‘master’ trunks.

In addition it is necessary to define from which of the ‘slave’ trunks the gateway recovers its clock:

- TDMBusPSTNAutoClockEnable = 1 (the gateway automatically selects one of the connected ‘slave’ trunks)

Or

- TDMBusLocalReference = # (Trunk index: 0 to 7, default = 0)

Notes:

- To configure the TDM Bus Clock Source parameters, refer to Section 5.6.4 on page 90.
- When the gateway is used in a ‘non-span’ configuration, the internal gateway clock must be used (as explained above).

10.2 ISDN Overlap Dialing

Overlap dialing is a dialing scheme used by several ISDN variants to send and / or receive called number digits one right after the other (or several at a time). As opposed to en-bloc dialing in which a complete number is sent.

The gateway can optionally support ISDN overlap dialing for incoming ISDN calls for the entire gateway by setting ‘ISDNRxOverlap’ to 1, or per E1/T1 span by setting ‘ISDNRxOverlap_x’ to 1 (‘x’ represents the number of the trunk, 0 to 7).

To play a Dial tone to the ISDN user side when an empty called number is received, set ‘ISDNINCallsBehavior = 65536’ (bit #16) causing the Progress Indicator to be included in the SetupAck ISDN message.

The gateway stops collecting digits (for ISDN→IP calls) when:

- The sending device transmits a ‘sending complete’ IE in the ISDN Setup or the following INFO messages to signal that no more digits are going to be sent.
- The inter-digit timeout (configured by the parameter ‘TimeBetweenDigits’) expires. The default for this timeout is 4 seconds.
- The maximum allowed number of digits (configured by the parameter ‘MaxDigits’) is reached. The default is 30 digits.
- A match is found with the defined digit map (configured by the parameter, DigitMapping).

Relevant parameters (described in Table 6-11 on page 189):

- ISDNRxOverlap
- ISDNRxOverlap_x
- TimeBetweenDigits
- MaxDigits
- ISDNInCallsBehavior
- DigitMapping

10.3 Using ISDN NFAS

In regular (non-NFAS) T1 ISDN trunks, a single 64 kbps channel carries signaling for the other 23 B-channels of that particular T1 trunk. This channel is called the D-channel and usually resides on timeslot # 24.

The ISDN Non-Facility Associated Signaling (NFAS) feature enables use of a single D-channel to control multiple PRI interfaces.

With NFAS it is possible to define a group of T1 trunks, called an NFAS group, in which a single D-channel carries ISDN signaling messages for the entire group. The NFAS group's B-channels are used to carry traffic, such as voice or data. The NFAS mechanism also enables definition of a backup D-channel on a different T1 trunk, to be used if the primary D-channel fails.

The NFAS group comprises several T1 trunks. Each T1 trunk is called an 'NFAS member'. The T1 trunk whose D-channel is used for signaling is called the 'Primary NFAS Trunk'. The T1 trunk whose D-channel is used for backup signaling is called the 'Backup NFAS Trunk'. The primary and backup trunks each carry 23 B-channels while all other NFAS trunks each carry 24 B-channels.

The gateway supports multiple NFAS groups. Each group should contain different T1 trunks.

The NFAS group is identified by an NFAS GroupID number (possible values are 1, 2, 3 and 4). To assign a number of T1 trunks to the same NFAS group, use the parameter 'NFASGroupNumber_x = groupID'. 'x' stands for the physical trunkID (0 to 7).

The parameter 'DchConfig_x = Trunk_type' is used to define the type of NFAS trunk. Trunk_type is set to 0 for the primary trunk, to 1 for the backup trunk and to 2 for an ordinary NFAS trunk. 'x' stands for the physical trunkID (0 to 7).

For example, to assign the first four gateway T1 trunks to NFAS group #1, in which trunk #0 is the primary trunk and trunk #1 is the backup trunk, use the following configuration:

```
NFASGroupNumber_0 = 1
NFASGroupNumber_1 = 1
NFASGroupNumber_2 = 1
NFASGroupNumber_3 = 1
DchConfig_0 = 0            ;Primary T1 trunk
DchConfig_1 = 1            ;Backup T1 trunk
DchConfig_2 = 2            ;24 B-channel NFAS trunk
DchConfig_3 = 2            ;24 B-channel NFAS trunk
```

The NFAS parameters are described in Table 6-11 on page 189.

Note: In the current version the NFAS parameters cannot be configured via the 'Trunk Settings' screen in the Embedded Web Server. Use *ini* file configuration instead.

10.3.1 NFAS Interface ID

Several ISDN switches require an additional configuration parameter per T1 trunk that is called 'Interface Identifier'. In NFAS T1 trunks the Interface Identifier is sent explicitly in Q.931 Setup / Channel Identification IE for all NFAS trunks, except for the B-channels of the Primary trunk (refer to note 1 below).

The Interface ID can be defined per each member (T1 trunk) of the NFAS group, and must be coordinated with the configuration of the Switch.

The default value of the Interface ID is identical to the number of the physical T1 trunk (0 for the first gateway trunk, 1 for the second gateway T1 trunk etc. up to 7).

To define an explicit Interface ID for a T1 trunk (that is different from the default), use the following parameters:

- ISDNIBehavior_x = 512 (x = 0 to 7 identifying the gateway's physical trunk)
- ISDNNFASInterfaceID_x = ID (x = 0 to 255)

Notes:

- Usually the Interface Identifier is included in the Q.931 Setup/Channel Identification IE only on T1 trunks that doesn't contain the D-channel. Calls initiated on B-channels of the Primary T1 trunk, by default, don't contain the Interface Identifier. Setting the parameter 'ISDNIBehavior_x' to 2048' forces the inclusion of the Channel Identifier parameter also for the Primary trunk.
- The parameter 'ISDNNFASInterfaceID_x = ID' can define the 'Interface ID' for any Primary T1 trunk, even if the T1 trunk is not a part of an NFAS group. However, to include the Interface Identifier in Q.931 Setup/Channel Identification IE configure 'ISDNIBehavior_x = 2048' in the *ini* file.

10.3.2 Working with DMS-100 Switches

The DMS-100 switch requires the following NFAS Interface ID definitions:

- InterfaceID #0 for the Primary trunk
- InterfaceID #1 for the Backup trunk
- InterfaceID #2 for a 24 B-channel T1 trunk
- InterfaceID #3 for a 24 B-channel T1 trunk
- Etc.

For example, if four T1 trunks on a gateway's are configured as a single NFAS group that is used with a DMS-100 switch, the following parameters should be used:

```
ISDNNFASInterfaceID_0 = 0
ISDNNFASInterfaceID_1 = 2
ISDNNFASInterfaceID_2 = 3
ISDNNFASInterfaceID_3 = 4
NFASGroupNumber_0 = 1
NFASGroupNumber_1 = 1
NFASGroupNumber_2 = 1
NFASGroupNumber_3 = 1
DchConfig_0 = 0 ;Primary T1 trunk
DchConfig_2 = 2 ;24 B-channel NFAS trunk
DchConfig_3 = 2 ;24 B-channel NFAS trunk
DchConfig_4 = 2 ;24 B-channel NFAS trunk
```

10.4 Redirect Number and Calling Name (Display)

The following tables define the gateway's redirect number and calling name (Display) support for various PRI variants:

Table 10-1: Calling Name (Display)

	DMS-100	NI-2	4/5ESS	Euro ISDN
NT→TE	Yes	Yes	Yes	Yes
TE→NT	Yes	Yes	Yes	No

Table 10-2: Redirect Number

	DMS-100	NI-2	4/5ESS	Euro ISDN
NT→TE	Yes	Yes	Yes	Yes
TE→NT	Yes	Yes	Yes	No

11 Advanced System Capabilities

11.1 Restoring Networking Parameters to their Initial State

You can use the 'Reset' button to restore the Mediant 2000 / TP-1610 networking parameters (described in Table 4-1) to their factory default values and to reset the username and password.

Note that the gateway returns to the software version burned in flash. This process also restores the gateway's parameters to their factory settings. Therefore, you must load your previously backed-up *ini* file, or the default *ini* file (received with the software kit) to set them to their correct values.

This option is currently supported (for Mediant 2000 and TP-1610) on one media gateway module (trunks 1-8) only.

➢ **To restore the networking parameters of the Mediant 2000 / TP-1610 to their initial state, take these 6 steps:**

1. Disconnect the Mediant 2000 from the power and network cables.
2. Reconnect the power cable; the gateway is powered up. After approximately 45 seconds the ACT LED blinks for about 4 seconds.
3. While the ACT LED is blinking, press shortly on the reset button (located on the front panel); the gateway resets a second time and is restored with factory default parameters (username: 'Admin', password: 'Admin').
4. Reconnect the network cable.
5. Assign the Mediant 2000 IP address (refer to Section 4.2 on page 50).
6. Load your previously backed-up *ini* file, or the default *ini* file (received with the software kit). To load the *ini* file via the Embedded Web Server, refer to Section 5.6.6 on page 96.

11.2 Establishing a Serial Communications Link with the Mediant 2000

Use serial communication software (e.g., HyperTerminal™) to establish a serial communications link with the Mediant 2000 via the RS-232 connection. You can use this link to change the networking parameters (Section 4.2.3 on page 51) and to receive error / notification messages.

Note: The TP-260 amd TP-1610 don't provide an RS-232 port.

➢ To establish a serial communications link with the Mediant 2000 via the RS-232 port, take these 2 steps:

1. Connect the RS-232 port to your PC (refer to Section 3.1.4.3 on page 41.
2. Use a serial communication software (e.g., HyperTerminal™) with the following communications port settings:
   - Baud Rate: 115,200 bps
   - Data bits: 8
   - Parity: None
   - Stop bits: 1
   - Flow control: None

Note that after resetting the gateway, the information, shown in Figure 11-1 below, appears on the terminal screen. This information can be used to determine possible Mediant 2000 initialization problems, such as incorrectly defined (or undefined) local IP address, subnet mask, etc.

Figure 11-1: RS-232 Status and Error Messages

```
MAC address = 00-90-8F-01-00-9E
Local IP address = 10.1.37.6
Subnet mask = 255.255.0.0
Default gateway IP address = 10.1.1.5
TFTP server IP address = 10.1.1.167
Boot file name = ram35136.cmp
INI file name = gateway.ini
Call agent IP address = 10.1.1.18
Log server IP address = 0.0.0.0
Full/Half Duplex state = HALF DUPLEX
Flash Software Burning state = OFF
Serial Debug Mode = OFF
Lan Debug Mode = OFF
BootLoad Version 1.75
Starting TFTP download... Done.
Mediant 2000 Version 4.60.00
```

11.3 Automatic Update Mechanism

The gateway is capable of automatically updating its *cmp*, *ini* and configuration files. These files can be stored on any standard Web, FTP or NFS server/s and can be loaded periodically to the gateway via HTTP, HTTPS, FTP or NFS. This mechanism can be used even for Customer Premise(s) Equipment (CPE) devices that are installed behind NAT and firewalls.

The Automatic Update mechanism is applied separately to each file. For the detailed list of available files and their corresponding parameters, refer to Table 6-2 on page 138.

Note: The Automatic Update mechanism assumes the external Web server conforms to the HTTP standard. If the Web server ignores the If-Modified-Since header, or doesn't provide the current date and time during the HTTP 200 OK response, the gateway may reset itself repeatedly. To overcome this problem, adjust the update frequency (AutoUpdateFrequency).

Three methods are used to activate the Automatic Update mechanism:

- After the gateway starts-up (refer to the Startup process described in Figure 11-3).
- At a configurable time of the day (e.g., 18:00). This option is disabled by default.
- At fixed intervals (e.g., every 60 minutes). This option is disabled by default.

The following *ini* file example can be used to activate the Automatic Update mechanism.

Figure 11-2: Example of an *ini* File Activating the Automatic Update Mechanism

```
# DNS is required for specifying domain names in URLs
DnsPriServerIP = 10.1.1.11

# Load an extra configuration ini file using HTTP
IniFileURL = 'http://webserver.corp.com/AudioCodes/inifile.ini'
# Load Call Progress Tones file using HTTPS
CptFileUrl = 'https://10.31.2.17/usa_tones.dat'
# Load Voice Prompts file using FTPS with user ‘root’ and password ‘wheel’
VPFileUrl = 'ftps://root:wheel@ftpserver.corp.com/vp.dat'

# Update every day at 03:00 AM
AutoUpdatePredefinedTime = '03:00'
# Note: The cmp file isn’t updated since it is disabled by default
(AutoUpdateCmpFile).
```

Refer to the following notes:

- When HTTP or HTTPS are used, the gateway contacts the Web server/s and queries for the requested files. The *ini* file is loaded only if it was modified since the last automatic update. The *cmp* file is loaded only if its version is different from the version stored on the gateway’s non-volatile memory. All other auxiliary files (e.g., CPT) are updated only once. To update a previously-loaded auxiliary file, you must update the parameter containing its URL.
- To load different configurations (*ini* files) for specific gateways, add the string ‘<MAC>’ to the URL. This mnemonic is replaced with the gateway's hardware MAC address. Resulting in an *ini* file name request that contains the gateway’s MAC address.
- To automatically update the *cmp* file, use the parameter ‘CmpFileURL’ to specify its name and location. As a precaution (to protect the gateway from an accidental update), the Automatic Update mechanism doesn’t apply to the *cmp* file by default. Therefore, (to enable it) set the parameter ‘AutoUpdateCmpFile’ to 1.

The following example illustrates how to utilize Automatic Updates for deploying devices with minimum manual configuration.

➢ To utilize Automatic Updates for deploying the gateway with minimum manual configuration, take these 5 steps:

1. Set up a Web server (in the following example it is http://www.corp.com/) where all configuration files are to be stored.
2. To each device, pre-configure the following parameter (DHCP / DNS are assumed): IniFileURL = 'http://www.corp.com/master_configuration.ini'
3. Create a file named master_configuration.ini, with the following text:

```
# Common configuration for all devices
# ------------------------------------
CptFileURL = 'http://www.corp.com/call_progress.dat'
# Check for updates every 60 minutes
AutoUpdateFrequency = 60

# Additional configuration per device
# -----------------------------------
# Each device loads a file named after its MAC address,
# (e.g., config_00908F033512.ini)
IniFileURL = 'http://www.corp.com/config_<MAC>.ini'

# Reset the device after configuration is updated.
# The device resets after all of the files are processed.
ResetNow = 1
```

   You can modify the master_configuration.ini file (or any of the config_<MAC>.ini files) at any time. The gateway queries for the latest version every 60 minutes and applies the new settings immediately.

4. For additional security, use HTTPS or FTPS. The gateway supports HTTPS (RFC 2818) and FTPS using the AUTH TLS method <draft-murray-auth-ftp-ssl-16> for the Automatic Update mechanism.
5. To load configuration files from an NFS server, the NFS file system parameters should be defined in the configuration *ini* file. The following is an example of an *ini* file for loading files from NFS servers using NFS version 2.

```
# Define NFS servers for Automatic Update
[ NFSServers ]
FORMAT NFSServers_Index = NFSServers_HostOrIP, NFSServers_RootPath,
NFSServers_NfsVersion;
NFSServers 1 = 10.31.2.10, /usr/share, 2 ;
NFSServers 2 = 192.168.100.7, /d/shared, 2 ;
[ \NFSServers ]

CptFileUrl = 'file://10.31.2.10/usr/share/public/usa_tones.dat'
VpFileUrl = 'file://192.168.100.7/d/shared/audiocodes/voiceprompt.dat'
```

11.4 Startup Process

The startup process (illustrated in Figure 11-3 on page 252) begins when the gateway is reset (physically or from the Web / SNMP) and ends when the operational software is running. In the startup process, the network parameters, software and configuration files are obtained.

After the gateway powers up or after it is physically reset, it broadcasts a BootRequest message to the network. If it receives a reply (from a BootP server), it changes its network parameters (IP address, subnet mask and default gateway address) to the values provided. If there is no reply from a BootP server and if DHCP is enabled (DHCPEnable = 1), the gateway initiates a standard DHCP procedure to configure its network parameters.

After changing the network parameters, the gateway attempts to load the *cmp* and various configuration files from the TFTP server's IP address, received from the BootP/DHCP servers. If a TFTP server's IP address isn't received, the gateway attempts to load the software (*cmp*) file and / or configuration files from a preconfigured TFTP server (refer to Section 11.3 on page 249). Thus, the gateway can obtain its network parameters from BootP or DHCP servers and its software and configuration files from a different TFTP server (preconfigured in *ini* file).

If BootP/DHCP servers are not found or when the gateway is reset from the Web / SNMP, it retains its network parameters and attempts to load the software (*cmp*) file and / or configuration files from a preconfigured TFTP server.

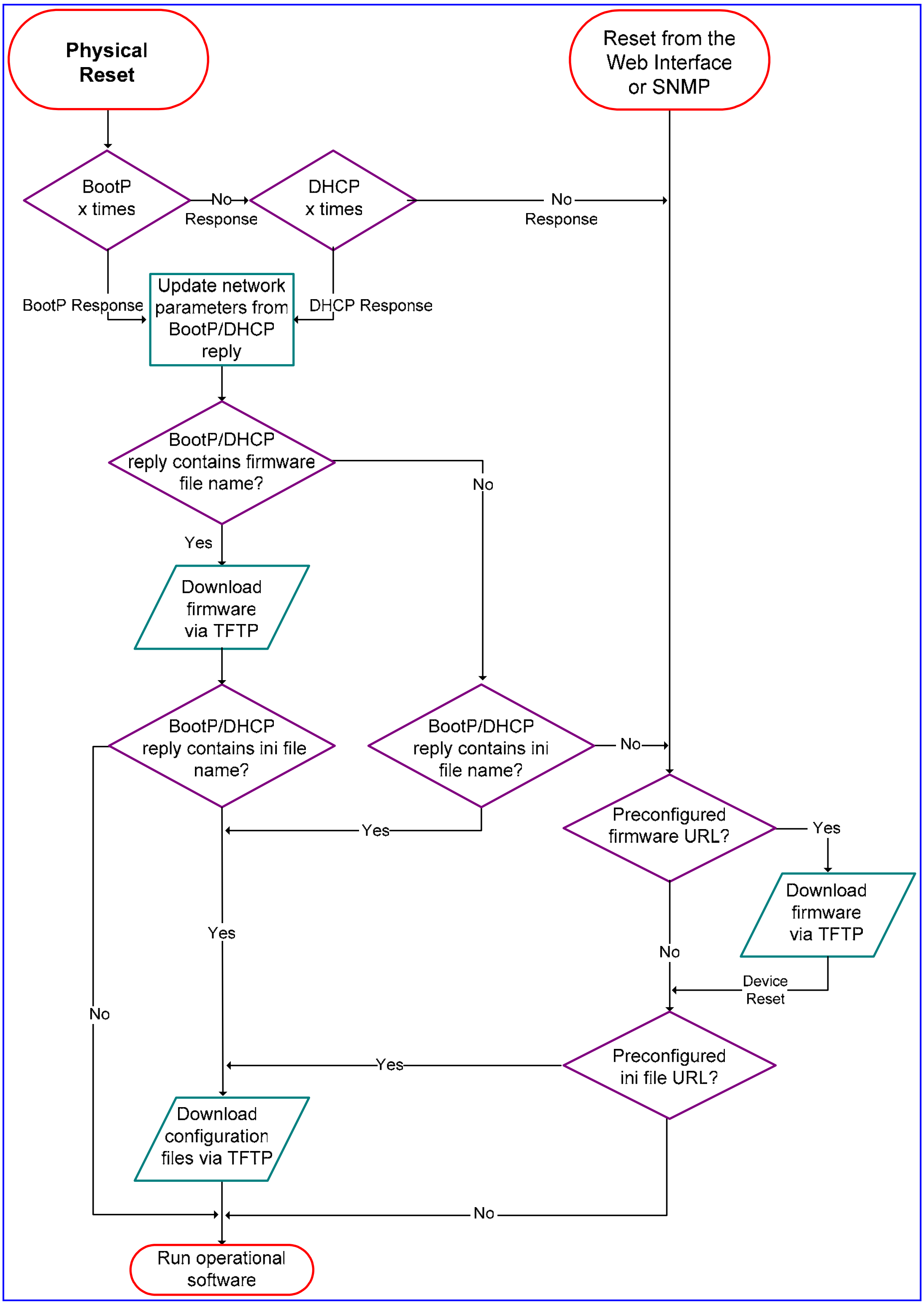
If a preconfigured TFTP server doesn't exist, the gateway operates using the existing software and configuration files loaded on its non-volatile memory.

Note that after the operational software runs, if DHCP is configured, the gateway attempts to renew its lease with the DHCP server.

Notes:

- Though DHCP and BootP servers are very similar in operation, the DHCP server includes some differences that could prevent its operation with BootP clients. However, many DHCP servers, such as Windows™ NT DHCP server, are backward-compatible with BootP protocol and can be used for gateway configuration.
- The time duration between BootP/DHCP requests is set to 1 second by default. This can be changed by the BootPDelay *ini* file parameter. Also, the number of requests is 3 by default and can be changed by BootPRetries *ini* file parameter. (Both parameters can also be set using the BootP command line switches).

Figure 11-3: Gateway's Startup Process



11.5 Using Parameter Tables

The gateway uses parameter tables to group related parameters of specific entities and manage them together. These tables, similar to regular parameters, can be configured via the *ini* file, Embedded Web Server, SNMP, etc.

Tables are composed of lines and columns. Columns represent parameters' types. Lines represent specific entities. The instances in each line are called line attributes. Lines in table may represent (for example) a trunk, an NFS file system, list of timers for a given application, etc.

Table 11-1 and Table 11-2 below provide useful examples for reference.

Table 11-1: Example of Parameter Table - Remote Management Connections

Index Fields: 1. Connection Number				
Connection Number	**User Name**	**User Password**	**Time Connected (msec)**	**Permissions**
0	Admin	Yellow9	0	All
1	Gillian	Red5	1266656	Read Only
2	David	Orange6	0	Read Write

Table 11-2: Example of Parameter Table - Port-to-Port Connections

Index Fields: 1. Source Ports 2. Destination IP 3. Destination Port				
Source Port	**Destination IP**	**Destination Port**	**Connection Name**	**Application Type**
2020	10.4.1.50	2020	ATM_TEST_EQ	LAB_EQ
2314	212.199.201.20	4050	ATM_ITROP_LOOP	LAB_EQ
6010	10.3.3.41	6010	REMOTE_MGMT	MGMT

Note: Table 11-1 and Table 11-2 are provided as examples for the purpose of illustration only and are not actually implemented in the gateway.

11.5.1 Table Indices

Each line in a table must be unique. Therefore, each table defines one or more Index fields. The combination of the Index fields determines the 'line-tag'. Each line-tag appears only once.

In the example provided in Table 11-1 there is only one Index field. This is the simplest way to mark lines.

In the example provided in Table 11-2 there are three Index fields. This more complicated method is a result of the application it represents.

11.5.2 Table Permissions

Each column has a 'permission' attribute that is applied to all instances in the column. This permission determines if and when a field can be modified. Several permissions can be applied to each column.

The following permissions are available:

- **Read:** Value of the field can be read.
- **Write:** Value of the field can be modified.
- **Create:** Value for the field must be provided at creation time (the default values, set to all fields, determine the initial values).
- **Maintenance Write:** The value of the field can only be modified when the entity represented by the line is in maintenance state (each table includes rules that determine when it is in maintenance state).

In the example in Table 11-1 it is assumed that the columns ‘User Name’ and ‘User Password’ have Read-Create permissions. The column ‘Time Connected’ has a Read permission, and the column ‘Permissions’ has Read-Create-Maintenance Write permissions.

11.5.3 Dynamic Tables vs. Static Tables

- **Static Tables:** Static tables don’t support adding new lines or removing (deleting) existing lines. All lines in a Static table are pre-configured with default values. Users can only modify the values of the existing lines. After reset, all lines in a Static table are available.
- **Dynamic Tables:** Dynamic tables support adding and removing lines. They are always initialized as empty tables with no lines. Users should add lines to a Dynamic table via the *ini* file or at run-time via the Embedded Web Server for example.

Note: Certain dynamic tables may initialize a line (or more) at start-up. If so, it is explained in the specific table's documentation.

11.5.4 Secret Tables

A table is defined as a secret table if it contains at least a single secret data field or if it depends on another secret table. A secret data field is a field that mustn’t be revealed to the user. For example, in the IPSec application, IPSec tables are defined as secret tables as the IKE table contains a pre-shared key that must be concealed. Therefore, the SPD table that depends on the IKE table is defined as a secret table as well.

There are two major differences between tables and secret tables:

- The secret field itself cannot be viewed via SNMP, Web or any other application.
- *ini* file behavior: Secret tables are never displayed in an uploaded *ini* file (e.g., when performing a 'Get *ini* File from Web' operation). Instead, there is a commented title that states that the secret table is present at the gateway and is not to be revealed. Secret tables are always kept in the gateway’s non-volatile memory and can be over-written by new tables that are provided in a new *ini* file. If a secret table appears in an *ini* file, it replaces the current table regardless of its content. To delete a secret table from the gateway, provide an empty table of the same type (with no data lines) as part of a new *ini* file; the empty table replaces the previous table in the gateway.

11.5.5 Using the *ini* File to Configure Parameter Tables

You can use the *ini* file to add / modify parameter tables. When using tables, Read-Only parameters are not loaded, as they cause an error when trying to reload the loaded file. Therefore, Read-Only parameters mustn't be included in tables in the *ini* file. Consequently, tables are loaded with all parameters having at least one of the following permissions: Write, Create or Maintenance Write.

Parameter tables (in an uploaded *ini* file) are grouped according to the applications they configure (e.g., NFS, IPSec). When loading an *ini* file to the gateway, the recommended policy is to include only tables that belong to applications that are to be configured (Dynamic tables of other applications are empty, but static tables are not).

The *ini* file includes a Format line that defines the columns of the table to be modified (this may vary from *ini* file to *ini* file for the same table). The Format line must only include columns that can be modified (parameters that are not specified as Read-Only).

An exception is Index-fields that are always mandatory. In the example provided in Table 11-1, all fields except for the 'Time Connected' field are loaded.

11.5.5.1 Structure of Parameter Tables in the *ini* File

Tables are composed of four elements:

- The title of the table - The name of the table in square brackets (e.g., [MY_TABLE_NAME]).
- A Format line - Specifies the columns of the table (by their string names) that are to be configured.
  - The first word of the Format line must be 'FORMAT', followed by the names of the Indices fields, and an equal sign '='. After the equal sign the names of the columns are listed.
  - Items must be separated by a comma ','.
  - The Format line must end with a semicolon ';'.
- Data line(s) – Contain the actual values of the parameters. The values are interpreted according to the Format line. The first word of the Data line must be the table's string name followed by the Index fields.
  - Items must be separated by a comma ','.
  - A Data line must end with a semicolon ';'.
- End-of-Table-Mark - Indicates the end of the table. The same string used for the table's title, preceded by a forward slash '\' (e.g., [\MY_TABLE_NAME]).

Figure 11-4 displays an example of the structure of a parameter table in the *ini* file.

Figure 11-4: Structure of a Parameter Table in the *ini* File

```
; Table: Items Table.
; Fields: Item_Name, Item_Serial_Number, Item_Color, Item_weight.
; NOTE: Item_Color is not specified. It will be given default value.
[Items_Table]
; Fields declaration
Format Item_Index =  Item_Name, Item_Serial_Number, Item_weight;
Items_Table 0 = Computer, 678678, 6;
Items_Table 6 = Computer-screen, 127979, 9;
Items_Table 2 = Computer-pad, 111111, $$;
[\Items_Table]
```

Refer to the following notes:

- Indices (in both the Format and the Data lines) must appear in the same order determined by the specific table's documentation. The Index field must never be omitted.
- The Format line can include a sub-set of the configurable fields in a table. In this case, all other fields are assigned with the pre-defined default values for each configured line.
- The order of the fields in the Format line isn't significant (as opposed to the Index-fields). The fields in the Data lines are interpreted according to the order specified in the Format line.
- The sign '$$' in a Data line indicates that the user wants to assign the pre-defined default value to it.
- The order of the Data lines is insignificant.
- Data lines must match the Format line, i.e., it must contain exactly the same number of Indices and Data fields and must be in exactly the same order.
- A line in a table is identified by its table-name and Index fields. Each such line may appear only once in the *ini* file.
- Table dependencies:
  Certain tables may depend on other tables. For example, one table may include a field that specifies an entry in another table. This method is used to specify additional attributes of an entity, or to specify that a given entity is part of a larger entity. The tables must appear in the order of their dependency (i.e., if Table X is referred to by Table Y, Table X must appear in the *ini* file before Table Y).

11.6 Customizing the Web Interface

Customers incorporating the gateway's into their portfolios can customize the device's Web Interface to suit their specific corporate logo and product naming conventions.

Customers can customize the Web Interface's title bar (AudioCodes' title bar is shown in Figure 11-5; a customized title bar is shown in Figure 11-6).

Figure 11-5: User-Customizable Web Interface Title Bar

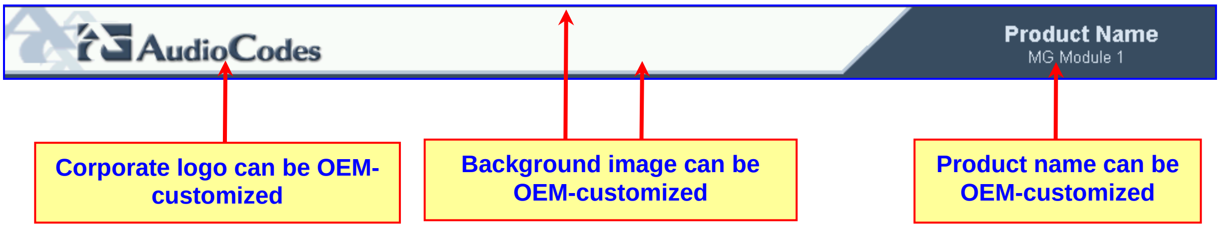


Figure 11-6: Customized Web Interface Title Bar

➢ **To customize the title bar via the Web Interface, take these 3 steps:**

1. Replace the main corporate logo (refer to Section 11.6.1 below).
2. Replace the title bar's background image file (refer to Section 11.6.2 on page 259).
3. Customize the product's name (refer to Section 11.6.3 on page 260).

11.6.1 Replacing the Main Corporate Logo

The main corporate logo can be replaced either with a different logo image file (refer to Section 11.6.1.1 below) or with a text string (refer to Section 11.6.1.2 on page 259). Note that when the main corporation logo is replaced, AudioCodes' logo on the left bar (refer to Figure 5-2) and in the Software Upgrade Wizard (refer to Section 5.8.1 on page 115) disappear.

Also note that the browser's title bar is automatically updated with the string assigned to the WebLogoText parameter when AudioCodes' default logo is not used.

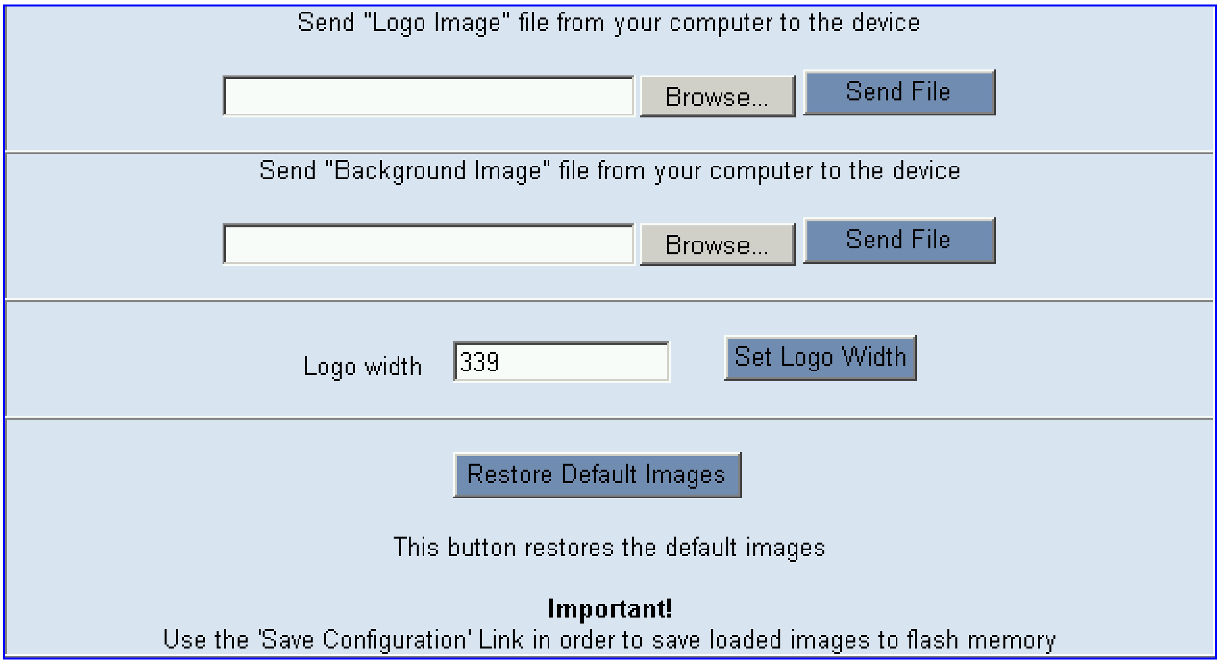
11.6.1.1 Replacing the Main Corporate Logo with an Image File

Note: Use a gif, jpg or jpeg file for the logo image. It is important that the image file has a fixed height of 59 pixels (the width can be configured). The size of the image files (logo and background) is limited each to 64 kbytes.

➢ **To replace the default logo with your own corporate image via the Web Interface, take these 7 steps:**

1. Access the gateway's Embedded Web Server (refer to Section 5.3 on page 58).
2. In the URL field, append the suffix 'AdminPage' (note that it's case-sensitive) to the IP address, e.g., http://10.1.229.17/AdminPage.
3. Click **Image Load to Device**; the Image Download screen is displayed (shown in Figure 11-7).

Figure 11-7: Image Download Screen



4. Click the **Browse** button in the 'Send Logo Image File from your computer to the Device' box. Navigate to the folder that contains the logo image file you want to load.
5. Click the **Send File** button; the file is sent to the device. When loading is complete, the screen is automatically refreshed and the new logo image is displayed.
6. Note the appearance of the logo. If you want to modify the width of the logo (the default width is 339 pixels), in the 'Logo Width' field, enter the new width (in pixels) and click the **Set Logo Width** button.
7. To save the image to flash memory so it is available after a power fail, refer to Section 5.9.2 on page 124.

The new logo appears on all Web Interface screens.

Tip: If you encounter any problem during the loading of the files, or you want to restore the default images, click the **Restore Default Images** button.

➢ To replace the default logo with your own corporate image via the *ini* file, take these 2 steps:

1. Place your corporate logo image file in the same folder as where the device's *ini* file is located (i.e., the same location defined in the BootP/TFTP configuration utility). For detailed information on the BootP/TFTP, refer to Appendix D on page 353.
2. Add/modify the two *ini* file parameters in Table 11-3 according to the procedure described in Section 6.2 on page 127.

Note that loading the device's *ini* file via the 'Configuration File' screen in the Web Interface doesn't load the corporate logo image files as well.

Table 11-3: Customizable Logo *ini* File Parameters

Parameter	Description
LogoFileName	The name of the image file containing your corporate logo. Use a gif, jpg or jpeg image file. The default is AudioCodes' logo file. **Note:** The length of the name of the image file is limited to 47 characters.
LogoWidth	Width (in pixels) of the logo image. **Note:** The optimal setting depends on the resolution settings. The default value is 339, which is the width of AudioCodes' displayed logo.

11.6.1.2 Replacing the Main Corporate Logo with a Text String

The main corporate logo can be replaced with a text string.

- To replace AudioCodes' default logo with a text string *via the Web Interface*, modify the two *ini* file parameters in Table 11-4 according to the procedure described in Section 11.6.4 on page 261.
- To replace AudioCodes' default logo with a text string *via the ini file*, add/modify the two *ini* file parameters in Table 11-4 according to the procedure described in Section 6.2 on page 127.

Table 11-4: Web Appearance Customizable *ini* File Parameters

Parameter	Description
UseWebLogo	0 = Logo image is used (default). 1 = Text string is used instead of a logo image.
WebLogoText	Text string that replaces the logo image. The string can be up to 15 characters.

11.6.2 Replacing the Background Image File

The background image file is duplicated across the width of the screen. The number of times the image is duplicated depends on the width of the background image and screen resolution. When choosing your background image, keep this in mind.

Note: Use a gif, jpg or jpeg file for the background image. It is important that the image file has a fixed height of 59 pixels. The size of the image files (logo and background) is limited each to 64 kbytes.

➢ To replace the background image via the Web, take these 6 steps:

1. Access the gateway's Embedded Web Server (refer to Section 5.3 on page 58).
2. In the URL field, append the suffix 'AdminPage' (note that it's case-sensitive) to the IP address, e.g., http://10.1.229.17/AdminPage.
3. Click the **Image Load to Device**, the Image Download screen is displayed (shown in Figure 11-7).
4. Click the **Browse** button in the 'Send Background Image File from your computer to gateway' box. Navigate to the folder that contains the background image file you want to load.
5. Click the **Send File** button; the file is sent to the device. When loading is complete, the screen is automatically refreshed and the new background image is displayed.
6. To save the image to flash memory so it is available after a power fail, refer to Section 5.9.2 on page 124.

The new background appears on all Web Interface screens.

Tip 1: If you encounter any problem during the loading of the files, or you want to restore the default images, click the **Restore Default Images** button.

Tip 2: When replacing both the background image and the logo image, first load the logo image followed by the background image.

➢ To replace the background image via the *ini* file, take these 2 steps:

1. Place your background image file in the same folder as where the device's *ini* file is located (i.e., the same location defined in the BootP/TFTP configuration utility). For detailed information on the BootP/TFTP, refer to Appendix D on page 353.
2. Add/modify the *ini* file parameters in Table 11-5 according to the procedure described in Section 6.2 on page 127.

Note that loading the device's *ini* file via the 'Configuration File' screen in the Web Interface doesn't load the logo image file as well.

Table 11-5: Customizable Logo *ini* File Parameters

Parameter	Description
BkgImageFileName	The name (and path) of the file containing the new background. Use a gif, jpg or jpeg image file. The default is AudioCodes background file. **Note:** The length of the name of the image file is limited to 47 characters.

11.6.3 Customizing the Product Name

The Product Name text string can be modified according to OEMs specific requirements.

- To replace AudioCodes' default product name with a text string *via the Web Interface*, modify the two *ini* file parameters in Table 11-6 according to the procedure described in Section 11.6.4 on page 261.
- To replace AudioCodes' default product name with a text string *via the ini file*, add/modify the two *ini* file parameters in Table 11-6 according to the procedure described in Section 6.2 on page 127.

Table 11-6: Web Appearance Customizable *ini* File Parameters

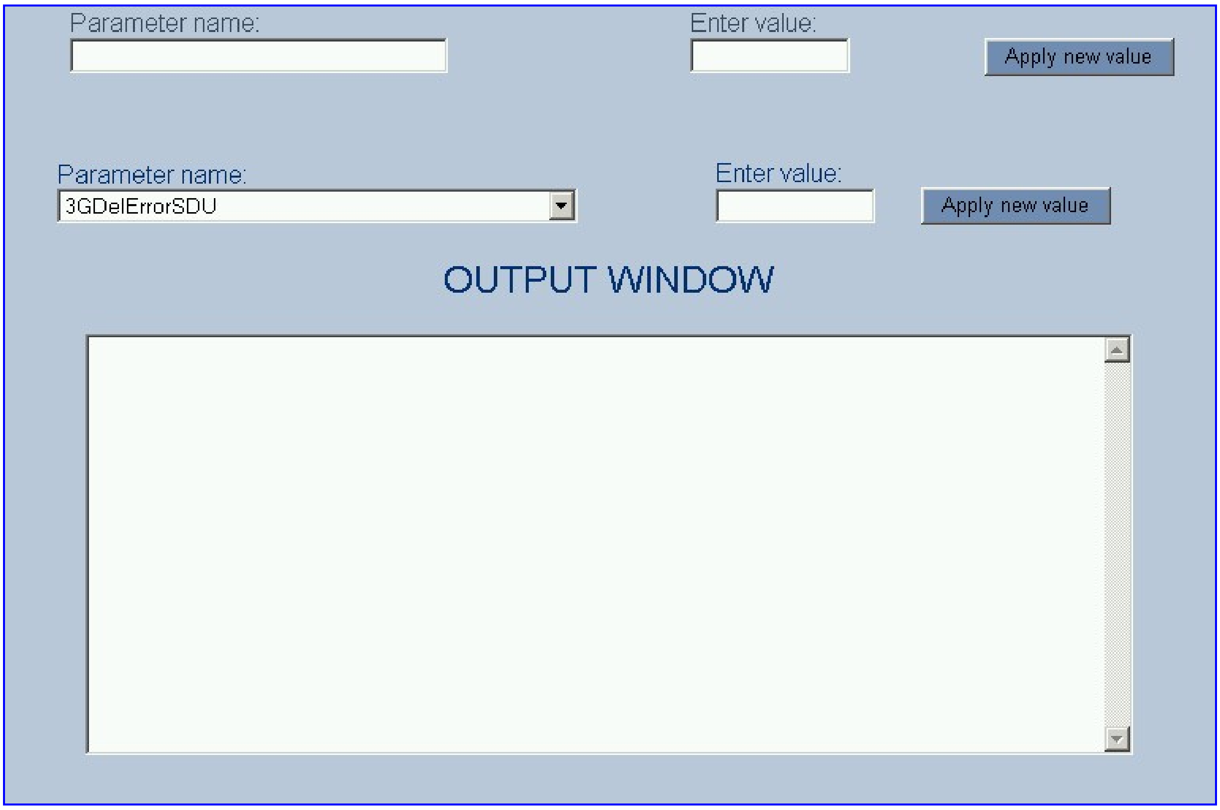
Parameter	Description
UseProductName	0 = Don't change the product name (default). 1 = Enable product name change.
UserProductName	Text string that replaces the product name. The default is 'Mediant 2000'. The string can be up to 29 characters.

11.6.4 Modifying *ini* File Parameters via the Web AdminPage

➢ **To modify *ini* file parameters via the AdminPage, take these 6 steps:**

1. Access the gateway's Embedded Web Server (refer to Section 5.3 on page 58).
2. In the URL field, append the suffix 'AdminPage' (note that it's case-sensitive) to the IP address, e.g., http://10.1.229.17/AdminPage.
3. Click the **INI Parameters** option, the INI Parameters screen is displayed (shown in Figure 11-8).

Figure 11-8: INI Parameters Screen



4. In the **Parameter Name** dropdown list, select the required *ini* file parameter.
5. In the **Enter Value** field to the right, enter the parameter's new value.
6. Click the **Apply new value** button to the right; the INI Parameters screen is refreshed, the parameter name with the new value appears in the fields at the top of the screen and the **Output Window** displays a log displaying information on the operation.

Note: You cannot load the image files (e.g., logo/background image files) to the device by choosing a file name parameter in this screen.

11.7 Software Upgrade Key

The gateways are supplied with a Software Upgrade Key already pre-configured for each of its TrunkPack Modules (TPM).

Users can later upgrade their gateway's features, capabilities and quantity of available resources by specifying what upgrades they require, and purchasing a new key to match their specification.

The Software Upgrade Key is sent as a string in a text file, to be loaded onto the gateway. Stored in the device's non-volatile flash memory, the string defines the features and capabilities allowed by the specific key purchased by the user. The device allows users to utilize *only these* features and capabilities. A new key overwrites a previously installed key.

> **Note:** The Software Upgrade Key is an encrypted key. Each TPM utilizes a unique key. The Software Upgrade Key is provided by AudioCodes only.

11.7.1 Backing up the Current Software Upgrade Key

Back up your current Software Upgrade Key before loading a new key to the device. You can always reload this backed-up key to restore your device capabilities to what they originally were if the 'new' key doesn't comply with your requirements.

➢ **To backup the current Software Upgrade Key, take these 5 steps:**

1. Access the devices Embedded Web Server (refer to Section 5.3 on page 58).
2. Click the **Software Update** button.
3. Click the **Software Upgrade Key** tab; the Software Upgrade Key screen is displayed (shown in Figure 11-9).
4. Copy the string from the 'Current Key' field and paste it in a new file.
5. Save the text file with a name of your choosing.

11.7.2 Loading the Software Upgrade Key

After receiving the Software Upgrade Key file (do not modify its contents in any way), ensure that its first line is [LicenseKeys] and that it contains one or more lines in the following format:

S/N<Serial Number of TPM> = <long Software Upgrade Key>

For example: S/N370604 = jCx6r5tovCIKaBBbhPtT53Yj...

One S/N must match the S/N of your device. The device's S/N can be viewed in the 'Device Information' screen (refer to Section 5.7.4 on page 113).

You can load a Software Upgrade Key using:

- The Embedded Web Server (refer to Section 11.7.2.1).
- The BootP/TFTP configuration utility (refer to Section 11.7.2.2 on page 264).
- AudioCodes' EMS (refer to Section 15.10 on page 326 and to AudioCodes' EMS User's Manual or EMS Product Description).

11.7.2.1 Loading the Software Upgrade Key Using the Embedded Web Server

➢ **To load a Software Upgrade Key using the Web Server, take these 6 steps:**

1. Access the devices Embedded Web Server (refer to Section 5.3 on page 58).
2. Click the **Software Update** button.
3. Click the **Software Upgrade Key** tab; the Software Upgrade Key screen is displayed (shown in Figure 11-9).
4. When loading a single key S/N line to a device:

   Open the Software Upgrade Key file (it should open in Notepad), select and copy the key string of the device's S/N and paste it into the Web field 'New Key'. If the string is sent in the body of an email, copy and paste it from there. Click the **Add Key** button.
5. When loading a Software Upgrade Key text file containing multiple S/N lines to a device (refer to Figure 11-10):

   Click the **Browse** button in the 'Send "Upgrade Key" file from your computer to the device' field, and navigate to the Software Upgrade Key text file. Click the **Send File** button. The new key is loaded to the device, validated and if valid is burned to memory. The new key is displayed in the 'Current Key' field.

   Validate the new key by scrolling through the 'Key features:' panel and verifying the presence / absence of the appropriate features.
6. After verifying that the Software Upgrade Key was successfully loaded, reset the device; the new capabilities and resources are active.

Figure 11-9: Software Upgrade Key Screen

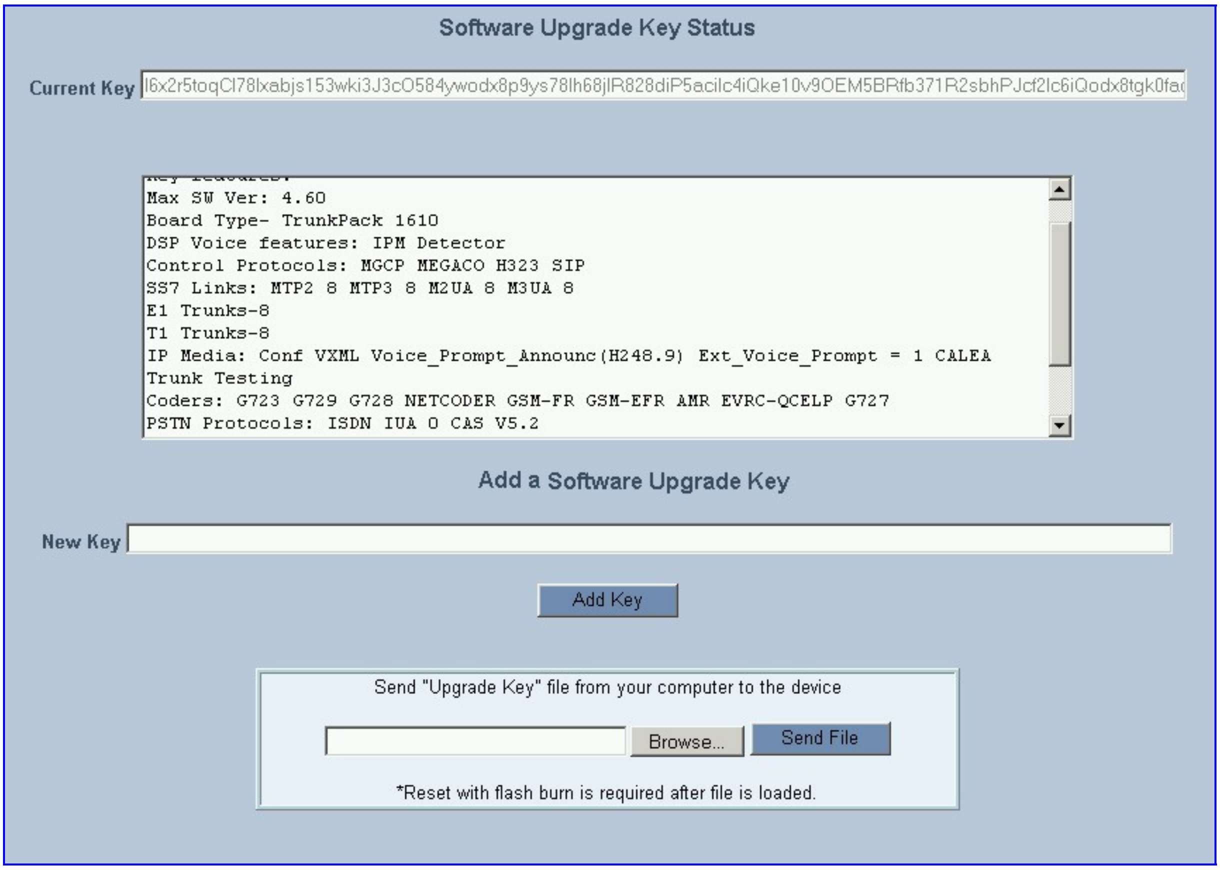


Figure 11-10: Example of a Software Upgrade Key File Containing Multiple S/N Lines

```
sample.ini - Notepad
File  Edit  Format  Help
[LicenseKeys]
;Board Type 29
S/N241182 =
okRTr5topwYMblZd4NN2a3Qhm4NJfidaagUyehso94APbBF85hF4by0cmQZlf2B8bMcze7JQ9kMSa5h641R1aOkeEb9AddF894Zx
S/N242519 = tmxTr5to0mlMblZdoPd2a3Qh9zJJfidafilyehsogOQPbBF8pjl4by0c9jxlf2B8eOoze7JQgywSa5h6o391aOkeTlIAddF8c6Fx
S/N226403 = tmxTr5to0lsMblZdoOB2a3Qh9yJJfidafilyehsogN4PbBF8piZ4by0c9ixlf2B8eOoze7JQgxgSa5h6o2x1aOkeTJlAddF8c6Fx
S/N226417 = r6xTr5to25sMblZdfiB2a3Qh5OJJfida92lyehsoix4PbBF8eOZ4by0c52xlf2B88yoze7JQiNgSa5h6fyx1aOkeXZlAddF8amFx
;Board Type 24
S/N241182 =
okRTr5topwYMblZd4NN2a3wkm4NJfidaagUyehso94APbBF85hF4by0cmQZlf2B8bMcze7JQ9kMSa5h641R1aOkeEb9AddF8938s
S/N242519 = tmxTr5to0mlMblZdoPd2a3wk9zJJfidafilyehsogOQPbBF8pjl4by0c9jxlf2B8eOoze7JQgywSa5h6o391aOkeTlIAddF8c1ss
S/N226403 = tmxTr5to0lsMblZdoOB2a3wk9yJJfidafilyehsogN4PbBF8piZ4by0c9ixlf2B8eOoze7JQgxgSa5h6o2x1aOkeTJlAddF8c1ss
S/N226417 = r6xTr5to25sMblZdfiB2a3wk5OJJfida92lyehsoix4PbBF8eOZ4by0c52xlf2B88yoze7JQiNgSa5h6fyx1aOkeXZlAddF8ahss
```

11.7.2.2 Loading the Software Upgrade Key Using BootP/TFTP

➢ **To load the Software Upgrade Key file using BootP/TFTP, take these 5 steps:**

1. Place the file in the same location you've saved the *device's cmp* file. Note that in order to load the Software Upgrade Key via TFTP server, the extension of the key file must be *ini*.
2. Start the BootP/TFTP configuration utility and from the **Services** menu in the main screen, choose option **Clients**; the Client Configuration screen is displayed (refer to Figure D-4 on page 359).
3. From the drop-down list in the **INI File** field, select the Software Upgrade Key file instead of the device's *ini* file. Note that the device's *cmp* file must be specified in the **Boot File** field.
4. Configure the initial BootP/TFTP parameters required, and click **OK** (refer to Appendix D on page 353).
5. Reset the device; the device's *cmp* and Software Upgrade Key files are loaded to the device.

11.7.3 Verifying that the Key was Successfully Loaded

After installing the key, you can determine in the Embedded Web Server's read-only 'Key features:' panel (**Software Update** menu > **Software Upgrade Key**) (refer to Figure 11-9) that the features and capabilities activated by the installed string match those that were ordered.

You can also verify that the key was successfully loaded to the device by accessing the Syslog server. For detailed information on the Syslog server, refer to Section 14.2 on page 304. When a key is successfully loaded, the following message is issued in the Syslog server:

'S/N___ Key Was Updated. The Board Needs to be Reloaded with *ini* file\n'

11.7.4 Troubleshooting an Unsuccessful Loading of a Key

If the Syslog server indicates that a Software Upgrade Key file was unsuccessfully loaded (the SN_ line is blank), take the following preliminary actions to troubleshoot the issue:

- Open the Software Upgrade Key file and check that the S/N line of the specific device whose key you want to update is listed in it. If it isn't, contact AudioCodes.
- Verify that you've loaded the correct file and that you haven't loaded the device's *ini* file or the CPT *ini* file by mistake. Open the file and ensure that the first line is [LicenseKeys].
- Verify that you didn't alter in any way the contents of the file.

11.7.5 Abort Procedure

Reload the key you backed-up in Section Backing up the Current Software Upgrade Key on page 262 to restore your device capabilities to what they originally. To load the backed-up key use the procedure described in Section Loading the Software Upgrade Key on page 262.

Reader's Notes

12 Special Applications

12.1 TDM Tunneling

The gateway's TDM Tunneling feature allows you to tunnel groups of digital trunk spans or timeslots (B-channels) over the IP network. TDM Tunneling utilizes the internal routing capabilities of the gateway (working without Proxy control) to receive voice and data streams from TDM (1 to 16 E1/T1/J1) spans or individual timeslots, convert them into packets and transmit them automatically over the IP network (using point-to-point or point-to-multipoint gateway distributions). A gateway opposite it (or several AudioCodes gateways, when point-to-multipoint distribution is used) converts the IP packets back into TDM traffic. Each timeslot can be targeted to any other timeslot within a trunk in the opposite gateway.

12.1.1 Implementation

When TDM Tunneling is enabled (‘EnableTDMOverIP’ is set to 1 on the originating gateway), the originating gateway automatically initiates H.323 calls from all enabled B-channels belonging to the E1/T1/J1 spans that are configured with the ‘Transparent’ protocol (for ISDN trunks), or ‘Ras CAS’ (for CAS trunks). The called number of each call is the internal phone number of the B-channel that the call originates from. The IP to Trunk Group routing table is used to define the destination IP address of the terminating gateway. The terminating gateway automatically answers these calls if its E1/T1 protocol is set to ‘Transparent’ (ProtocolType = 5), or ‘Raw CAS’ (ProtocolType = 3 for T1 and 9 for E1) and the parameter ‘ChannelSelectMode = 0’ (By Phone Number).

> **Note:** It is possible to configure both gateways to also operate in symmetric mode. To do so, set ‘EnableTDMOverIP’ to 1 and configure the Tel to IP Routing tables in both gateways. In this mode, each gateway (after it is reset) initiates calls to the second gateway. The first call for each B-channel is answered by the second gateway.

The gateway monitors the established connections continuously, if for some reason one or more calls are released, the gateway automatically reestablishes these ‘broken’ connections. In addition, when a failure in a physical trunk or in the IP network occurs, the gateways reestablish the tunneling connections as soon as the network restores.

> **Note:** It is recommended to use the keep-alive mechanism for each connection by activating ‘session expires’ timeout, and using Reinvite messages.

By utilizing the ‘Profiles’ mechanism (refer to Section 5.5.6 on page 77) you can configure the TDM Tunneling feature to choose different settings, based on a timeslot or groups of timeslots. For example, you can use low-bit-rate vocoders to transport voice, and ‘Transparent’ coder to transport data (e.g., for D-channel). You can also use Profiles to assign ToS (for DiffServ) per source, a time-slot carrying data or signaling gets a higher priority value than a time-slot carrying voice.

For tunneling of E1/T1 CAS trunks set the protocol type to Raw CAS (ProtocolType = 3 / 9) and enable RFC 2833 CAS relay mode (CASTransportType = 1).

Figure 12-1 and Figure 12-2 show an example of *ini* files for two Mediant 2000 gateways implementing TDM Tunneling for four E1 spans. Note that in this example both gateways are dedicated to TDM tunneling.

Figure 12-1: *ini* File Example for TDM Tunneling (Originating Side)

```
EnableTDMOverIP = 1

;E1_TRANSPARENT_31
ProtocolType_0 = 5
ProtocolType_1 = 5
ProtocolType_2 = 5
ProtocolType_3 = 5

prefix = '*,10.8.24.12' ;(IP address of the Mediant 2000 in the opposite
location)

; Channel selection by Phone number
ChannelSelectMode = 0

;Profiles can be used do define different coders per B-channels, such as
Transparent
; coder for B-channels (time slot 16) that carries PRI signaling.
TrunkGroup = 0/1-31,1000,1
TrunkGroup = 1/1-31,2000,1
TrunkGroup = 2/1-31,3000,1
TrunkGroup = 3/1-31,4000,1
TrunkGroup = 0/16-16,7000,2
TrunkGroup = 1/16-16,7001,2
TrunkGroup = 2/16-16,7002,2
TrunkGroup = 3/16-16,7003,2

CoderName = 'g7231'
CoderName = 'Transparent'

CoderName_1 = 'g7231'
CoderName_2 = 'Transparent'

TelProfile_1 = voice,$$,1,$$,$$,$$,$$,$$,$$,$$
TelProfile_2 = data,$$,2,$$,$$,$$,$$,$$,$$,$$
```

Figure 12-2: *ini* File Example for TDM Tunneling (Terminating Side)

```
;E1_TRANSPARENT_31
ProtocolType_0 = 5
ProtocolType_1 = 5
ProtocolType_2 = 5
ProtocolType_3 = 5

; Channel selection by Phone number
ChannelSelectMode = 0

TrunkGroup = 0/1-31,1000,1
TrunkGroup = 1/1-31,2000,1
TrunkGroup = 2/1-31,3000,1
TrunkGroup = 3/1-31,4000,1
TrunkGroup = 0/16-16,7000,2
TrunkGroup = 1/16-16,7001,2
TrunkGroup = 2/16-16,7002,2
TrunkGroup = 3/16-16,7003,2

CoderName = 'g7231'
CoderName = 'Transparent'
CoderName_1 = 'g7231'
CoderName_2 = 'Transparent'
TelProfile_1 = voice,$$,1,$$,$$,$$,$$,$$,$$,$$
TelProfile_2 = data,$$,2,$$,$$,$$,$$,$$,$$,$$
```

12.2 SS7 Tunneling

The Signaling System 7 (SS7) tunneling feature facilitates peer-to-peer transport of SS7 links between gateways that support AudioCodes' unique MTP2 (Message Transfer Part) Tunneling application (M2TN) for transferring SS7 MTP2 link data over IP. In this scenario, both sides of the link are pure TDM switches and are unaware of the IP tandem that is utilized between them. Using M2TN, the network operator can support SS7 connections over IP, carrying MTP level 3, as well as higher level SS7 layers (e.g., user parts and application protocols, such as TUP (Telephone User Part), Integrated ISUP (Services User Part), SCCP (Signaling Connection Control Part), TCAP (Transaction Capabilities Application Part)).

M2TN uses standard protocols, such as SIGTRAN (RFC 2719 Architectural Framework for Signaling Transport), SCTP (RFC 2960, Stream Control Transmission Protocol), M2UA (RFC 3331, MTP2 User Adaptation Layer), the latter being used for transporting SS7-MTP2 signaling information over IP. M2UA architecture is shown in Figure 12-3. M2TN architecture is shown in Figure 12-4.

Figure 12-3: M2UA Architecture

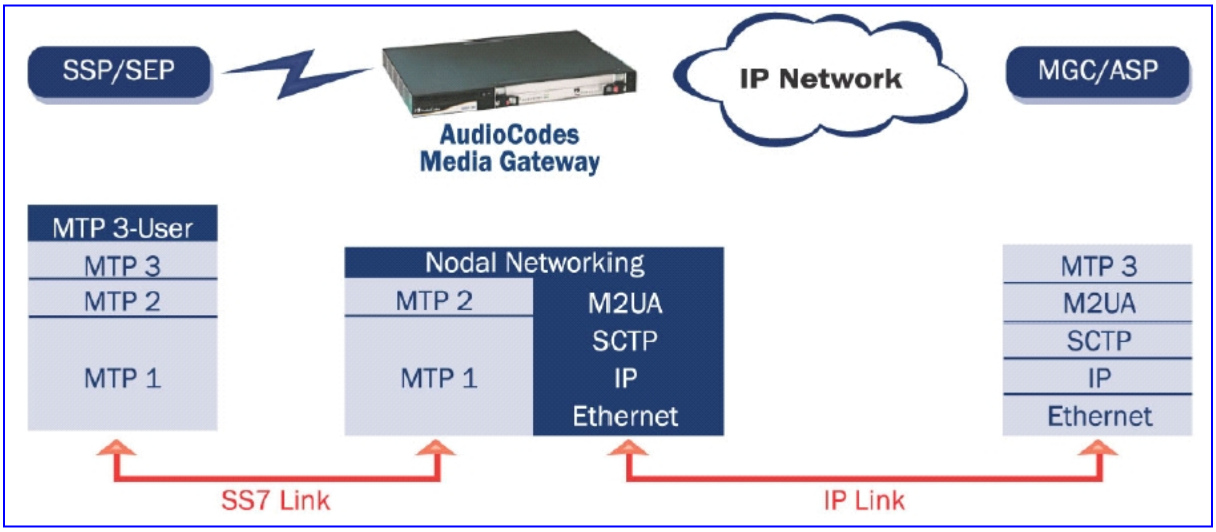
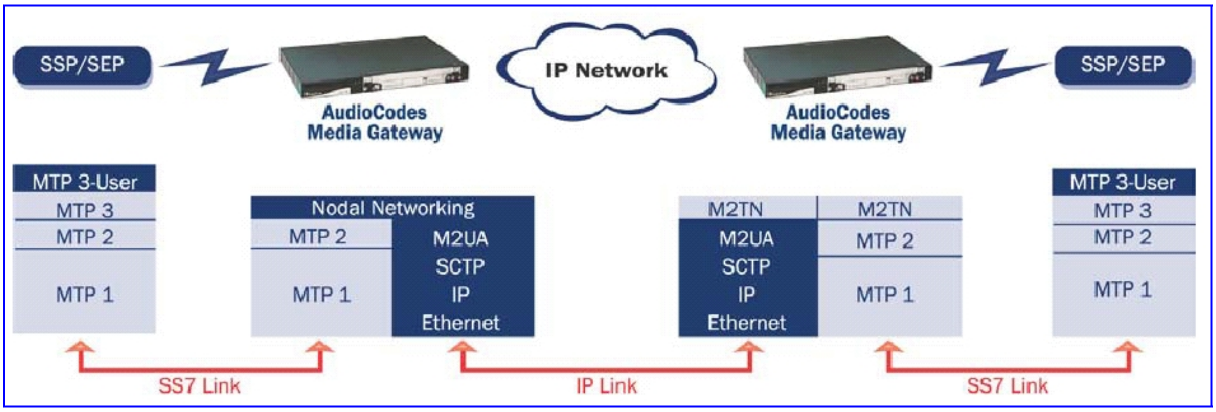


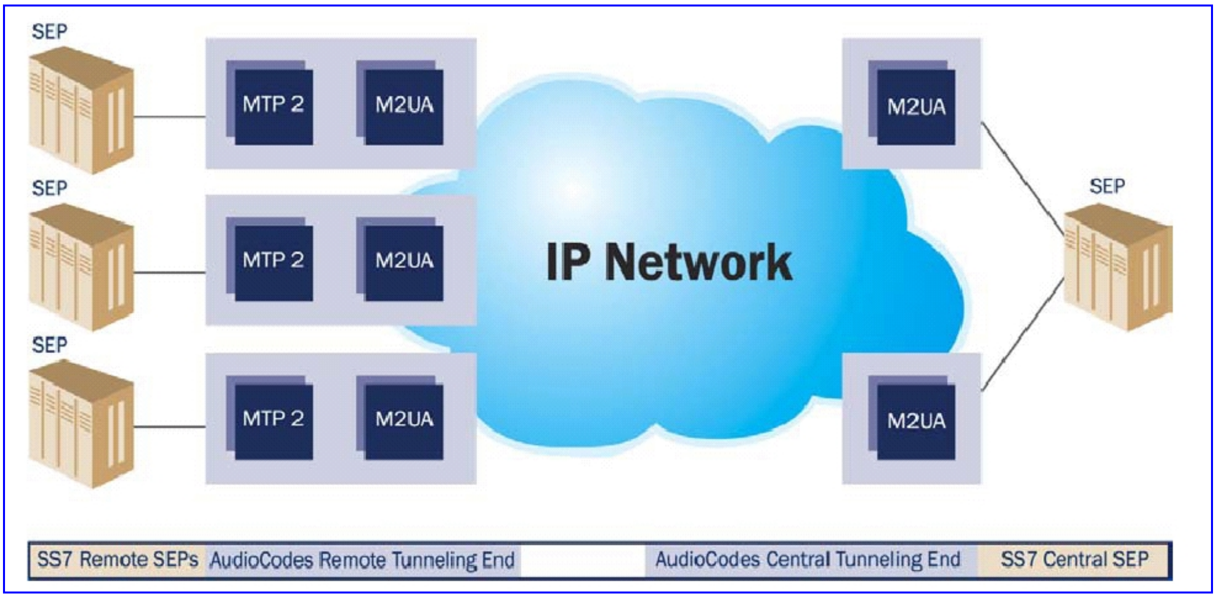
Figure 12-4: M2TN Architecture



12.2.1 MTP2 Tunneling Technology

The SS7 tunneling technology is based on a pairing of remote and central gateways, as shown in Figure 12-5. The remote gateways are configured to backhaul MTP layer 2 signaling over the IP network using standard M2UA protocol (over SCTP protocol). The function of the M2TN entity is to transmit traffic and handle all management events between MTP2 on the TDM side and M2UA's MGC (Media Gateway Controller) entity on the IP side. Only the actual SS7 MSU (Message Signaling Unit) data is sent. Management of the SS7 link is performed using M2UA without transporting the MTP2 LSSU (Link Status Signaling Unit) and FISU (Fill in Signaling Unit) messages over IP. These messages, in addition to MTP2 timing, are terminated and supported, respectively, by the remote and central sides. Therefore, the MTP2 connections are not affected by the fact that they are transported over IP.

Figure 12-5: Protocol Architecture for MTP2 Tunneling



12.2.2 SS7 Characteristics

- Only standard protocols are used on external interfaces (MTP2 on PSTN side, and M2UA over SCTP on IP side) - the M2TN application resides internally on the gateway.
- No extra signaling point codes are required; both endpoints are unaware that the SS7 connection is via IP.
- Several links from multiple SS7 nodes can be concentrated into a single board on the ‘Central’ side (using several SCTP associations per gateway).
- The gateways can handle SS7 MTP2 tunneling and voice concurrently (does not require additional gateway or other server).
- Voice and signaling can be transferred on the same E1/T1 trunk (F-Links).
- IP traffic can be monitored via standard sniffing tools (e.g. protocol analyzers).

Note: Channels that are used for SS7 Tunneling mustn’t be defined in the Trunk Group table.

12.2.3 SS7 Parameters

The parameters in Table 12-1 below configure all MTP attributes simultaneously. To set each MTP attribute individually, add _xx (xx equals the element number in the range of 0 to 2) to the end of the *ini* file field name.

Table 12-1: SS7 Parameters (continues on pages 271 to 272)

ini File Parameter Name	Description
SS7_MTP2_Param_AERM_TIE [AERM TIE]	Defines the SS7 alignment emergency error rate threshold. The valid range is 0 to 10. The default value is 1.
SS7_MTP2_Param_AERM_TIN [AERM TIN]	Defines the SS7 alignment normal error rate threshold. The valid range is 0 to 20. The default value is 4.
SS7_MTP2_Param_Error_Correction_Method [Error Correction Method]	Defines the SLI error correction method. 0 = Basic (default) B = Basic P = PCR (Preventive Cyclic Retransmission)
SS7_MTP2_Param_IAC_CP [IAC CP]	Defines the number of aborted proving attempts before sending an out-of-service to MTP-3. The valid range is 0 to 10. The default value is 5.
SS7_MTP2_Param_Link_Rate [Link Rate]	Defines the SS7 SLI Link Rate. Choose either: 0 = 64 kbps (default) A = 64 kbps D = 56 kbps
SS7_MTP2_Param_LSSU_Length [LSSU Length]	Defines the SS7 MTP2 LSSU length as 1 or 2 (bytes). The valid range is 1 to 2. The default value is 1.
SS7_MTP2_Param_Octet_Counting [Octet Conting]	Defines the SS7 MTP2 Octet received while the OCTET is in counting mode (# of Octets received - N Octets - while in Octet counting mode). The valid range is 0 to 256. The default value is 16.
SS7_MTP2_Param_SUERM_SU_D [SUERM SU D]	Defines the SS7 Signal Unit error rate monitor D threshold. The valid range is 0 to 256. The default value is 256.
SS7_MTP2_Param_SUERM_T [SUERM T]	Defines the SS7 SUERM (Signal Unit Error Rate Monitor) T threshold. The valid range is 0 to 256. The default value is 64.
SS7_MTP2_Param_Timer_T1 [T1]	Defines the SS7 MTP2 T1 alignment ready timer (in msecs). The valid range is 0 to 100000. The default value is 50000.
SS7_MTP2_Param_Timer_T2 [T2]	Defines the SS7 MTP2 T2 unaligned timer (in msecs). The valid range is 0 to 200000. The default value is 150000.
SS7_MTP2_Param_Timer_T3 [T3]	Defines the SS7 MTP2 T3 timer aligned. The valid range is 0 to 20000. The default value is 2000.
SS7_MTP2_Param_Timer_T4E [T4E]	Defines the SS7 MTP2 T4e Emergency proving period timer (msec). The valid range is 0 to 5000. The default value is 500.
SS7_MTP2_Param_Timer_T4N [T4N]	Defines the SS7 MTP2 T4n Nominal proving period timer. The valid range is 0 to 15000. The default value is 8200.

Table 12-1: SS7 Parameters (continues on pages 271 to 272)

ini File Parameter Name	Description
SS7_MTP2_Param_Timer_T5 [T5]	Defines the SS7 MTP2 Sending SIB timer. The valid range is 0 to 2400. The default value is 120.
SS7_MTP2_Param_Timer_T6 [T6]	Defines the SS7 MTP2 Remote Congestion timer (in msec). The valid range is 0 to 10000. The default value is 6000.
SS7_MTP2_Param_Timer_T7 [T7]	Defines the SS7 MTP2 excessive delay of the ack timer (in msec). The valid range is 0 to 5000. The default value is 2000.

12.2.4 SS7 Parameter Tables

For detailed information on parameter tables, refer to Section 11.5 on page 253.

12.2.4.1 SIGTRAN Interface Groups

Table 12-2: SIGTRAN Interface Groups (continues on pages 272 to 273)

ini File Parameter Name	Description
SS7_SIG_IF_GR_INDEX [Group Number]	Indicates the SS7 interface group index for a line. The valid range is 0 to 15.
SS7_IF_GR_ID [Group ID]	Determines the SS7 SIGTRAN interface group index, for a line. The valid range is 0 to 0xFFFF. The default value is 0xFFFE.
SS7_SIG_SG_MGC [UAL Group Number]	Determines the SS7 SIGTRAN interface group Signaling Gateway (SG) and Media Gateway Controller (MGC) option. The valid range is 77(MGC), 83(SG). The default value is 83.
SS7_SIG_LAYER [Group Layer]	Determines the SIGTRAN group layer (IUA/M2UA/M3UA). Choose either: 0 = no_layer (default) 1 = iua 2 = m2ua 3 = m3ua 4 = m2tunnel 5 = V5ua
SS7_SIG_TRAF_MODE [Group Traffic Mode]	Determines the SS7 SIGTRAN interface group traffic mode. The valid range is 1 to 3. The default value is 1.
SS7_SIG_T_REC [Tr - Group Recovery Timer]	Determines the SIGTRAN group T recovery. The valid range is 0 to 10000000. The default value is 2000.
SS7_SIG_T_ACK [Ta - Group Acknowledge Timer]	Determines the SIGTRAN group T Ack (in msec). The valid range is 0 to 10000000. The default value is 2000.
SS7_SIG_T_HB [Th - Group Heartbeat Timer]	Determines the SIGTRAN group T Hb (in msec). The valid range is 0 to 10000000. The default value is 2000.
SS7_SIG_MIN_ASP [Group Minimal ASP Number]	Determines the SIGTRAN group minimal Application Server Process (ASP) number (minimum = 1). The valid range is 1 to 10 The default value is 1.

Table 12-2: SIGTRAN Interface Groups (continues on pages 272 to 273)

ini File Parameter Name	Description
SS7_SIG_BEHAVIOUR [Group Behavior Field]	Determines the SIGTRAN group behavior bit. The valid range is 0 to 0xFFFFFFFE. The default value is 0.
SS7_LOCAL_SCTP_PORT [Group Local SCTP Port]	Determines the SIGTRAN group SCTP port. The valid range is 0 to 0xFFFE. The default value is 0Xfffe.
SS7_SIG_NETWORK [Group Network Variant]	Determines the SIGTRAN group Network (ITU, ANSI, CHINA). The valid range is 1 to 3. The default value is 1.
SS7_DEST_SCTP_PORT [Group Destination SCTP Port]	Determines the SIGTRAN group destination SCTP port. The valid range is 0 to 0xFFFE. The default value is 0xFFFE.
SS7_DEST_IP [Group Destination SCTP IP]	Determines the SIGTRAN group destination IP Address The valid range is 0 to 0xFFFFFFFE. The default value is 0.
SS7_MGC_MX_IN_STREAM [Inbound Streams Number]	Determines the SIGTRAN group maximum inbound stream. The valid range is 2 to 0xFFFE. The default value is 2.
SS7_MGC_NUM_OUT_STREAM [Outbound Streams Number]	Determines the SIGTRAN group's number of outbound streams. The valid range is 2 to 0xFFFE. The default value is 2.

12.2.4.2 SIGTRAN Interface IDs

Table 12-3: SIGTRAN Interface IDs

ini File Parameter Name	Description
SS7_SIG_IF_ID_INDEX [Interface Number	Determines the SS7 interface ID index, for a line. The valid range is 0 to 15. The default value is 1.
SS7_SIG_IF_ID_VALUE [Interface ID]	Determines the SIGTRAN interface ID value. The valid range is 0 to 0xFFFFFFFE. The default value is 0.
SS7_SIG_IF_ID_NAME [Interface ID Name]	Determines the SIGTRAN interface ID (text string). The default string is 'INT_ID'.
SS7_SIG_IF_ID_OWNER_GROUP [Owner Group]	Determines the SIGTRAN interface ID owner group. The valid range 0 to 0xFFFE. The default value is 0.
SS7_SIG_IF_ID_LAYER [Sigtran Layer Type]	Determines the SIGTRAN group layer (IUA/M2UA/M3UA). 0 = no_layer (default) 1 = iua 2 = m2ua 3 = m3ua 4 = m2tunnel 5 = V5ua
SS7_SIG_IF_ID_NAI [IF ID NAI]	Determines the SIGTRAN interface ID NAI. The valid range 0 to 0xFFFE. The default value is 0xFFFE.
SS7_SIG_M3UA_SPC [M3UA Local Point Code]	Determines the SIGTRAN M3UA SPC. The valid range 0 to 0xFFFFFFFE. The default value is 0.

12.2.4.3 SS7 Signaling Link

Table 12-4: SS7 Signaling Link **(continues on pages 274 to 275)**

ini File Parameter Name	Description
SS7_LINK_INDEX [Link Number]	Determines the index field for a line. The valid range is 0 to 7. The default value is 0.
SS7_LINK_ROWSTATUS	Determines the RowStatus field for a line. The valid range is acPARAMSET_ROWSTATUS_DOESNOTEXIST to acPARAMSET_ROWSTATUS_DESTROY. The default value is acPARAMSET_ROWSTATUS_DOESNOTEXIST.
SS7_LINK_ACTION	Determines the management field for actions. 0 = acSS7LINK_PS_ACTION_NONE (default) 1 = acSS7LINK_PS_ACTION_OFFLINE 2 = acSS7LINK_PS_ACTION_INSERVICE 3 = acSS7LINK_PS_ACTION_ACTIVATE 4 = acSS7LINK_PS_ACTION_DEACTIVATE 5 = acSS7LINK_PS_ACTION_INHIBIT 6 = acSS7LINK_PS_ACTION_UNINHIBIT
SS7_LINK_ACTION_RESULT	Determines the management field for actions result. The valid range is acPARAMSET_ACTION_RESULT_SUCCEEDED to acPARAMSET_ACTION_RESULT_FAILED. The default value is acPARAMSET_ACTION_RESULT_SUCCEEDED.
SS7_LINK_NAME [Name]	String name for link parameters The default string is 'LINK'.
SS7_LINK_OPERATIONAL_STATE [Operative State]	Determines the operational state of a signaling link. 0 = L3_OFFLINE (default) 1 = L3_BUSY, 2 = L3_INSERVICE
SS7_LINK_ADMINISTRATIVE_STATE [Administrative State]	Determines the administrative state of a signaling link. 0 = L3_OFFLINE (default) 2 = L3_INSERVICE
SS7_LINK_TRACE_LEVEL [Trace]	Determines the trace level of a signaling link (level 2). The valid range is 0 to 1. The default value is 0.
SS7_LINK_L2_TYPE [Layer 2 Type]	Determines the link layer type - defines level 2 media of signaling link. 0 = SS7_SUBLINK_L2_TYPE_NONE (default) 1 = SS7_SUBLINK_L2_TYPE_MTP2 2 = SS7_SUBLINK_L2_TYPE_M2UA_MGC 3 = SS7_SUBLINK_L2_TYPE_SAAL
SS7_LINK_L3_TYPE [Layer 3 Type]	Determines the link high layer type - defines level 3 or L2 high layer of signaling link. 0 = SS7_SUBLINK_L3_TYPE_NONE (default) 1 = SS7_SUBLINK_L3_TYPE_M2UA_SG 2 = SS7_SUBLINK_L3_TYPE_MTP3 3 = SS7_SUBLINK_L3_TYPE_MTP2_TUNNELING
SS7_LINK_TRUNK_NUMBER [Trunk Number]	Determines the trunk number of a signaling link (TDM). The valid range is 0 to 7. The default value is 0.
SS7_LINK_TIMESLOT_NUMBER [Timeslot Number]	Determines the time-slot number of a signaling link (TDM). The valid range is 0 to 31. The default value is 16.
SS7_LINK_MTC_BUSY [Local Busy]	Determines the link local busy indicator – if set, indicates link is busy due to local mtc action. The valid range is 0 to 1. The default value is 0.
SS7_LINK_INHIBITION [Inhibition]	Determines the link inhibit indicator - if set, indicates link is inhibited. The valid range is 0 to 1. The default value is L3_LINK_UNINHIBITED.

Table 12-4: SS7 Signaling Link (continues on pages 274 to 275)

ini File Parameter Name	Description
SS7_LINK_LAYER2_VARIANT [Variant]	Determines the variant (layer 2) of signaling link (TDM). 0 = NET_VARIANT_OTHER 1 = NET_VARIANT_ITU (default) 2 = NET_VARIANT_ANSI 3 = NET_VARIANT_CHINA
SS7_LINK_MTP2_ATTRIBUTES [MTP2 Attributes Index]	Determines the MTP2 attributes of signaling link (TDM). The valid range is 0 to MAX_C7_MTP2_PARAMS_INDEX. The default value is 3.
SS7_CONGESTION_LOW_MARK [Congestion Low Watermark]	Determines the link congestion low mark of signaling link (TDM). The valid range is 0 to 255. The default value is 5.
SS7_CONGESTION_HIGH_MARK [Congestion High Watermark]	Determines the link congestion high mark of signaling link (TDM). The valid range is 0 to 255. The default value is 20.
SS7_LINK_M2UA_IF_ID [Interface ID]	Determines the interface ID (M2UA) of signaling link. The valid range is 0 to 0xFFFFFFFF. The default value is 0.
SS7_LINK_GROUP_ID [Group ID]	Determines the group ID (M3UA) of signaling link. The valid range is 0 to 0xFFFF. The default value is 0.
SS7_LINK_TNL_MGC_LINK_NUMBER	Determines the MTP2 Tunneling: MGC link number (MTP2 \other side\ of signaling link. The valid range is 0 to 7. The default value is 0.
SS7_LINK_TNL_ALIGNMENT_MODE	Determines the MTP2 Tunneling: Alignment mode of signaling links in tunnel. 0 = M3B_ALIGNMENT_NORMAL 1 = M3B_ALIGNMENT_EMERGENCY (default)
SS7_LINK_TNL_CONGESTION_MODE	Determines the MTP2 Tunneling: Congestion mode of signaling links in tunnel. 0 = M3B_CONGESTION_ACCEPT (default) 1 = M3B_CONGESTION_DISCARD
SS7_LINK_TNL_WAIT_START_COMPLETE_TIMER	Determines the MTP2 Tunneling Timer: wait start complete. The valid range is 500 to 0xFFFFFFFF. The default value is 30000.
SS7_LINK_TNL_OOS_START_DELAY_TIMER	Determines the MTP2 Tunneling Timer: OOS start delay. The valid range is 500 to 0xFFFFFFFF. The default value is 5000.
SS7_LINK_TNL_WAIT_OTHER_SIDE_INSV_TIMER	Determines the MTP2 Tunneling Timer: wait other side inservice. The valid range is 500 to 0xFFFFFFFF. The default value is 30000.
SS7_LINKSET_SN_INDEX [SN Number]	Determines the first index field for line. The valid range is 0 to 1. The default value is 0.
SS7_LINKSET_LINKSET_INDEX [Link-set Number]	Determines the second index field for line. The valid range is 0 to 7. The default value is 0.
SS7_LINKSET_ROWSTATUS	Determines the RowStatusField for line. The valid range is acPARAMSET_ROWSTATUS_DOESNOTEXIST to acPARAMSET_ROWSTATUS_DESTROY. The default value is acPARAMSET_ROWSTATUS_DOESNOTEXIST.
SS7_LINKSET_ACTION	Determines the management field for actions. 0 = acSS7LINKSET_PS_ACTION_NONE (default) 1 = acSS7LINKSET_PS_ACTION_OFFLINE 2 = acSS7LINKSET_PS_ACTION_INSERVICE 3 = acSS7LINKSET_PS_ACTION_ACTIVATE 4 = acSS7LINKSET_PS_ACTION_DEACTIVATE
SS7_SN_ACTION_RESULT	Determines the management field for actions result. The valid range is acPARAMSET_ACTION_RESULT_SUCCEEDED to acPARAMSET_ACTION_RESULT_FAILED. The default value is acPARAMSET_ACTION_RESULT_SUCCEEDED.

12.2.5 SS7 MTP2 Tunneling *ini* File Example

For the SS7 MTP2 tunneling *ini* file example, note the following:

- The first *ini* file acts as an MTP2 tunneling central side (M2UA MGC links).
- There are eight SS7 links - four links of type: MTP2 MGC, and four links of type MTP2. Each pair of links (one MTP2 MGC and one MTP2) defines an MTP2 tunnel.
- There is one interface that is used for the M2UA MGC <=> M2UA SG (Signaling Gateway) connection.
- There are four interface IDs defined: one per link (M2UA MGC side).
- This file is intended for ITU link variant (E1 trunks).

➢ To load the example SS7 MTP2 tunneling *ini* files to gateways, take these 3 steps:

1. Load the *ini* file that is shown in Figure 12-6 to a tunnel central gateway (MTP2 MGC). Load the *ini* file that is shown in Figure 12-7 to a tunnel remote gateway (MTP2 SG); the MGC gateway connects (over IP) to the SG gateway. For information on loading an *ini* file to the gateway, refer to Section 6.2 on page 127.
2. In the MGC gateway, change the parameter 'SS7_DEST_IP' to the actual IP address of the M2UA SG gateway.
3. Change the value of the 'SyslogServerIP' parameter in the MGC and SG gateways to your Syslog server IP address.

Figure 12-6: SS7 MTP2 Tunneling *ini* File Example - MGC

```
[TDM BUS configuration]

; 1=aLaw  3=ulaw

PCMLawSelect=    1

;1 - internal, 3 - mvip, 4 - Network, 8 - h110a, 9 - h110b, 10 - Netref

TDMBusClockSource= 1

[Trunk Configuration]

;e1_euro_isdn=1 t1_isdn=2 ;e1_cas_r2=8 (8 for fcd); e1_trans_62=5

ProtocolType = 5

TraceLevel   = 0

; acCLOCK_MASTER_ON =1

CLOCKMASTER= 1

;acUSER_TERMINATION_SIDE = 0

TerminationSide = 1

;acEXTENDED_SUPER_FRAME=0

FramingMethod = 0

;acB8ZS = 0   2 for E1_CAS - FCD

LineCode = 0

[SS7]
```

Figure 12-6: SS7 MTP2 Tunneling *ini* File Example - MGC

```
SS7_MTP2_PARAM_TIMER_T1_0=50000

SS7_MTP2_PARAM_TIMER_T2_0=150000

SS7_MTP2_PARAM_TIMER_T3_0=1000

SS7_MTP2_PARAM_TIMER_T4E_0=500

SS7_MTP2_PARAM_TIMER_T4N_0=8200

SS7_MTP2_PARAM_TIMER_T5_0=100

SS7_MTP2_PARAM_TIMER_T6_0=3000

SS7_MTP2_PARAM_TIMER_T7_0=2000

[syslog]

SYSLOGSERVERIP = 168.100.0.1

ENABLESYSLOG = 1

;FORCEEXCEPTIONDUMP = 1

WATCHDOGSTATUS = 0

[ SS7_LINK_TABLE ]

FORMAT SS7_LINK_INDEX = SS7_LINK_NAME, SS7_LINK_TRACE_LEVEL,
SS7_LINK_ADMINISTRATIVE_STATE,SS7_LINK_L2_TYPE, SS7_LINK_L3_TYPE, SS7_LINK_GROUP_ID,
SS7_LINK_M2UA_IF_ID;

SS7_LINK_TABLE 1 = new_link_1, 0, 2, 2, 3, 4, 50;

SS7_LINK_TABLE 3 = new_link_3, 0, 2, 2, 3, 4, 12;

SS7_LINK_TABLE 5 = new_link_5, 0, 2, 2, 3, 4, 18;

SS7_LINK_TABLE 7 = new_link_7, 0, 2, 2, 3, 4,  1;

[ \SS7_LINK_TABLE ]

[ SS7_LINK_TABLE ]

FORMAT SS7_LINK_INDEX = SS7_LINK_NAME, SS7_LINK_TRACE_LEVEL,
SS7_LINK_ADMINISTRATIVE_STATE,SS7_LINK_L2_TYPE, SS7_LINK_L3_TYPE,
SS7_LINK_TRUNK_NUMBER,SS7_LINK_TIMESLOT_NUMBER,
SS7_LINK_LAYER2_VARIANT,SS7_LINK_MTP2_ATTRIBUTES,SS7_CONGESTION_LOW_M ARK,
SS7_CONGESTION_HIGH_MARK, SS7_LINK_TNL_MGC_LINK_NUMBER, SS7_LINK_TNL_ALIGNMENT_MODE,
SS7_LINK_TNL_CONGESTION_MODE, SS7_LINK_TNL_WAIT_START_COMPLETE_TIMER,
SS7_LINK_TNL_OOS_START_DELAY_TIMER, SS7_LINK_TNL_WAIT_OTHER_SIDE_INSV_TIMER;

SS7_LINK_TABLE 0 = new_link_0, 0, 2, 1, 3, 0, 15, 1, 0, 5, 50, 1, 1,  0, 30000, 5000,
30000;

SS7_LINK_TABLE 2 = new_link_2, 0, 2, 1, 3, 3, 12, 1, 0, 5, 50, 3, 1, 0, 30000, 5000, 30000;

SS7_LINK_TABLE 4 = new_link_4, 0, 2, 1, 3, 6,  7, 1, 0, 5, 50, 5, 1, 0, 30000, 5000, 30000;

SS7_LINK_TABLE 6 = new_link_6, 0, 2, 1, 3, 7, 3, 1, 0, 5, 50, 7, 1, 0, 30000, 5000, 30000;
```

Figure 12-6: SS7 MTP2 Tunneling *ini* File Example - MGC

```
[ \SS7_LINK_TABLE ]

[ SS7_SIG_IF_GROUP_TABLE ]

FORMAT SS7_SIG_IF_GR_INDEX = SS7_IF_GR_ID,SS7_SIG_SG_MGC, SS7_SIG_LAYER, SS7_SIG_TRAF_MODE,
SS7_SIG_T_REC, SS7_SIG_T_ACK, SS7_SIG_T_HB, SS7_SIG_MIN_ASP, SS7_SIG_BEHAVIOUR,
SS7_LOCAL_SCTP_PORT, SS7_SIG_NETWORK, SS7_DEST_SCTP_PORT, SS7_DEST_IP,
SS7_MGC_MX_IN_STREAM, SS7_MGC_NUM_OUT_STREAM;

SS7_SIG_IF_GROUP_TABLE   4   =   4,   77,   4,   1,   2000,   2000,   30000,   1,   0,   2904,
1,2904,168.100.0.2,3,3;

[ \SS7_SIG_IF_GROUP_TABLE ]

[ SS7_SIG_INT_ID_TABLE ]

FORMAT      SS7_SIG_IF_ID_INDEX      =      SS7_SIG_IF_ID_VALUE,      SS7_SIG_IF_ID_NAME,
SS7_SIG_IF_ID_OWNER_GROUP, SS7_SIG_IF_ID_LAYER, SS7_SIG_IF_ID_NAI, SS7_SIG_M3UA_SPC;

SS7_SIG_INT_ID_TABLE 7 = 50, BELFAST12, 4, 4, 1, 0;

SS7_SIG_INT_ID_TABLE 8 = 12, AMSTERDAM, 4, 4, 3, 0;

SS7_SIG_INT_ID_TABLE 9 = 18, ROTERDAM , 4, 4, 5, 0;

SS7_SIG_INT_ID_TABLE 10 = 1, GAUDA    , 4, 4, 7, 0;

[ \SS7_SIG_INT_ID_TABLE ]
```

Figure 12-7: SS7 MTP2 Tunneling ini File Example - SG

```
[TDM BUS configuration]

; 1=aLaw  3=ulaw

PCMLawSelect=    1

;1 - internal, 3 - mvip, 4 - Network, 8 - h110a, 9 - h110b, 10 - Netref

TDMBusClockSource= 1

[Trunk Configuration]

;e1_euro_isdn=1 t1_isdn=2 ;e1_cas_r2=8 (8 for fcd); e1_trans_62=5

ProtocolType = 5

TraceLevel   = 0

; acCLOCK_MASTER_ON =1

ClockMaster= 1

TerminationSide = 1

;acEXTENDED_SUPER_FRAME=0

FramingMethod = 0

;acB8ZS = 0   2 for E1_CAS - FCD

LineCode = 0

WATCHDOGSTATUS = 0

[ SS7_LINK_TABLE ]

FORMAT SS7_LINK_INDEX = SS7_LINK_NAME, SS7_LINK_TRACE_LEVEL,
SS7_LINK_ADMINISTRATIVE_STATE,SS7_LINK_L2_TYPE, SS7_LINK_L3_TYPE,
SS7_LINK_TRUNK_NUMBER,SS7_LINK_TIMESLOT_NUMBER,SS7_LINK_M2UA_IF_ID;

SS7_LINK_TABLE 0 = new_link_0, 0, 2, 1,1,  1, 15,50;

SS7_LINK_TABLE 1 = new_link_1, 0, 2, 1,1,  2, 12, 12;

SS7_LINK_TABLE 2 = new_link_2, 0, 2, 1, 1, 4,  7,18;

SS7_LINK_TABLE 3 = new_link_3, 0, 2, 1, 1, 5,  3,1;

[\SS7_LINK_TABLE]

[ SS7_SIG_IF_GROUP_TABLE ]

FORMAT SS7_SIG_IF_GR_INDEX = SS7_IF_GR_ID,SS7_SIG_SG_MGC, SS7_SIG_LAYER, SS7_SIG_TRAF_MODE,
SS7_SIG_T_REC, SS7_SIG_T_ACK, SS7_SIG_T_HB, SS7_SIG_MIN_ASP, SS7_SIG_BEHAVIOUR,
SS7_LOCAL_SCTP_PORT, SS7_SIG_NETWORK;

SS7_SIG_IF_GROUP_TABLE 4 =  4,83, 2, 1, 2000, 2000, 30000, 1, 0, 2904, 1;

[ \SS7_SIG_IF_GROUP_TABLE ]

[ SS7_SIG_INT_ID_TABLE ]

FORMAT      SS7_SIG_IF_ID_INDEX      =      SS7_SIG_IF_ID_VALUE,      SS7_SIG_IF_ID_NAME,
SS7_SIG_IF_ID_OWNER_GROUP, SS7_SIG_IF_ID_LAYER, SS7_SIG_IF_ID_NAI, SS7_SIG_M3UA_SPC;
```

Figure 12-7: SS7 MTP2 Tunneling ini File Example - SG

```
SS7_SIG_INT_ID_TABLE 7 = 50, BELFAST12, 4, 4, 0, 0;

SS7_SIG_INT_ID_TABLE 8 = 12, AMSTERDAM, 4, 4, 1, 0;

SS7_SIG_INT_ID_TABLE 9 = 18, ROTERDAM , 4, 4, 2, 0;

SS7_SIG_INT_ID_TABLE 10 = 1, GAUDA    , 4, 4, 3, 0;

[ \SS7_SIG_INT_ID_TABLE ]
```

12.3 QSIG Tunneling

The gateway supports QSIG tunneling over SIP according to <draft-elwell-sipping-qsig-tunnel-03>. This method enables all QSIG messages to be sent as raw data in corresponding SIP messages using a dedicated message body. This mechanism is useful for two QSIG subscribers (connected to the same / different QSIG PBX) to communicate with each other over an IP network. Tunneling is supported for both directions (Tel to IP and IP to Tel).

The term tunneling means that messages are transferred 'as is' to the remote side, without being converted (QSIG→SIP→QSIG). The advantage of tunneling over QSIG→SIP interworking is that by using interworking, QSIG functionality can only be partially achieved. When tunneling is used, all QSIG capabilities are supported, whereas the tunneling medium (the SIP network) does not need to process these messages.

12.3.1 Implementation

QSIG messages are transferred in SIP messages in a separate Multipurpose Internet Mail Extensions (MIME) body. Therefore, if a message contains more than one body (e.g., SDP and QSIG), multipart MIME must be used. The Content-Type of the QSIG tunneled message is 'application/QSIG'. In addition, the gateway adds a Content-Disposition header in the following format:

```
Content-Disposition: signal; handling=required.
```

- **Call setup (originating gateway):**
  The QSIG SETUP request is encapsulated in a SIP INVITE message without being altered. After the SIP INVITE request is sent, the gateway doesn't encapsulate the following QSIG message until a SIP 200 OK response is received. If the originating gateway receives a 4xx, 5xx or 6xx response, it disconnects the QSIG call with a 'no route to destination' cause.
- **Call setup (terminating gateway):**
  After the terminating gateway receives a SIP INVITE request with a Content-Type: application/QSIG, it sends the encapsulated QSIG SETUP message to the Tel side and sends a 200 OK response (no 1xx response is sent) to IP. The 200 OK response includes an encapsulated QSIG CALL PROCEEDING message (without waiting for a CALL PROCEEDING message from the Tel side). If tunneling is disabled and the incoming INVITE includes a QSIG body, a 415 response is sent.
- **Mid-call communication:**
  After the SIP connection is established, all QSIG messages are encapsulated in SIP INFO messages.
- **Call tear-down:**
  The SIP connection is terminated once the QSIG call is complete. The RELEASE COMPLETE message is encapsulated in the SIP BYE message that terminates the session.

To enable QSIG tunneling set the parameter EnableQSIGTunneling to 1 on both the originating and terminating gateways, and the parameter 'ISDNDuplicateQ931BuffMode' to 128 (duplicate all messages) (both parameters are described in Section 6.14 on page 172).

13 Security

This section describes the security mechanisms and protocols implemented on the gateway. The following list specifies the available security protocols and their objectives:

- IPSec and IKE protocols are part of the IETF standards for establishing a secured IP connection between two applications. IPSec and IKE are used in conjunction to provide security for control and management protocols but not for media (refer to Section 13.1 below).
- SSL (Secure Socket Layer) / TLS (Transport Layer Security) – The SSL / TLS protocols are used to provide privacy and data integrity between two communicating applications over TCP/IP. They are used to secure the following applications: SIP Signaling (SIPS), Web access (HTTPS) and Telnet access (refer to Section 13.2 on page 290).
- Secured RTP (SRTP) according to RFC 3711, used to encrypt RTP and RTCP transport (refer to Section 13.3 on page 294).
- RADIUS (Remote Authentication Dial-In User Service) - RADIUS server is used to enable multiple-user management on a centralized platform (refer to Section 13.4 page 295.
- Internal Firewall allows filtering unwanted inbound traffic (refer to Section 13.5 on page 298).

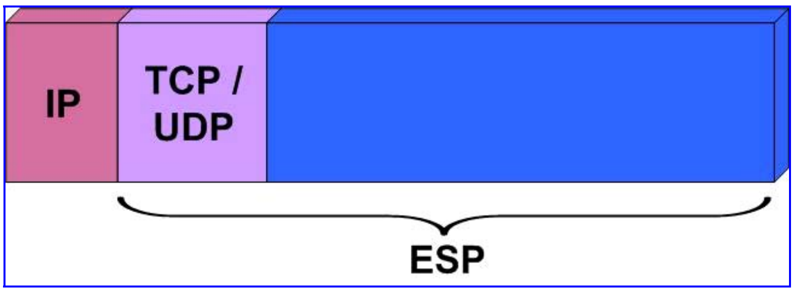
13.1 IPSec and IKE

IPSec and IKE protocols are part of the IETF standards for establishing a secured IP connection between two applications (also referred to as peers). Providing security services at the IP layer, IPSec and IKE are transparent to IP applications.

IPSec and IKE are used in conjunction to provide security for control and management (e.g., SNMP and Web) protocols but not for media (i.e., RTP, RTCP and T.38).

IPSec is responsible for securing the IP traffic. This is accomplished by using the Encapsulation Security Payload (ESP) protocol to encrypt the IP payload (illustrated in Figure 13-1 below). The IKE protocol is responsible for obtaining the IPSec encryption keys and encryption profile (known as IPSec Security Association (SA)).

Figure 13-1: IPSec Encryption



Note:	IPSec doesn't function properly if the gateway's IP address is changed on-the-fly due to the fact that the crypto hardware can only be configured on reset. Therefore, reset the gateway after you change its IP address.

13.1.1 IKE

IKE is used to obtain the Security Associations (SA) between peers (the gateway and the application it's trying to contact). The SA contains the encryption keys and profile used by the IPSec to encrypt the IP stream. The IKE table lists the IKE peers with which the gateway performs the IKE negotiation (up to 20 peers are available).

The IKE negotiation is separated into two phases: main mode and quick mode. The main mode employs the Diffie-Hellman (DH) protocol to obtain an encryption key (without any prior keys), and uses a pre-shared key to authenticate the peers. The created channel secures the messages of the following phase (quick mode) in which the IPSec SA properties are negotiated.

The IKE negotiation is as follows:

- Main mode (the main mode creates a secured channel for the quick mode)
  - SA negotiation – The peers negotiate their capabilities using four proposals. Each proposal includes three parameters: Encryption method, Authentication protocol and the length of the key created by the DH protocol. The key's lifetime is also negotiated in this stage. For detailed information on configuring the main mode proposals, refer to Section 13.1.3.1 on page 283.
  - Key exchange (DH) – The DH protocol is used to create a phase-1 key.
  - Authentication – The two peers authenticate one another using the pre-shared key (configured by the parameter 'IKEPolicySharedKey').
- Quick mode (quick mode negotiation is secured by the phase-1 SA)
  - SA negotiation – The peers negotiate their capabilities using four proposals. Each proposal includes two parameters: Encryption method and Authentication protocol. The lifetime is also negotiated in this stage. For detailed information on configuring the quick mode proposals, refer to the SPD table under Section 13.1.3.2 on page 286.
  - Key exchange – a symmetrical key is created using the negotiated SA.

IKE Specifications:

- Authentication mode - pre-shared key only
- Main mode is supported for IKE Phase 1
- Supported IKE SA encryption algorithms - Data Encryption Standard (DES), 3DES, and Advanced Encryption Standard (AES)
- Hash types for IKE SA - SHA1 and MD5

13.1.2 IPSec

IPSec is responsible for encrypting and decrypting the IP streams.

The IPSec Security Policy Database (SPD) table defines up to 20 IP peers to which the IPSec security is applied. IPSec can be applied to all packets designated to a specific IP address or to a specific IP address, port (source or destination) and protocol type.

Each outgoing packet is analyzed and compared to the SPD table. The packet's destination IP address (and optionally, destination port, source port and protocol type) are compared to each entry in the table. If a match is found, the gateway checks if an SA already exists for this entry. If it doesn't, the IKE protocol is invoked (refer to Section 13.1.1 above) and an IPSec SA is established. The packet is encrypted and transmitted. If a match isn't found, the packet is transmitted un-encrypted.

Note:	An incoming packet whose parameters match one of the entries of the SPD table but is received un-encrypted, is dropped.

IPSec specifications:

- Transport mode only.
- Encapsulation Security Payload (ESP) only.
- Support for Cipher Block Chaining (CBC).
- Supported IPSec SA encryption algorithms - DES, 3DES, and AES.
- Hash types for IPSec SA are SHA1 and MD5.

13.1.3 Configuring the IPSec and IKE

To enable IPSec and IKE on the gateway set the *ini* file parameter 'EnableIPSec' to 1.

13.1.3.1 IKE Configuration

The parameters described in Table 13-1 below are used to configure the first phase (main mode) of the IKE negotiation for a specific peer. A different set of parameters can be configured for each of the 20 available peers.

Table 13-1: IKE Table Configuration Parameters (continues on pages 283 to 284)

Parameter Name	Description
Shared Key **[IKEPolicySharedKey]**	Determines the pre-shared key (in textual format). Both peers must register the same pre-shared key for the authentication process to succeed. **Note 1:** The pre-shared key forms the basis of IPSec security and should therefore be handled cautiously (in the same way as sensitive passwords). It is not recommended to use the same pre-shared key for several connections. **Note 2:** Since the *ini* file is in plain text format, loading it to the gateway over a secure network connection is recommended, preferably over a direct crossed-cable connection from a management PC. For added confidentiality, use the encoded *ini* file option (described in Section 6.1 on page 127). **Note 3:** After it is configured, the value of the pre-shared key cannot be obtained via Web, *ini* file or SNMP (refer to Section 13.1.3.3 on page 289).
First to Fourth Proposal Encryption Type **[IKEPolicyProposalEncryption_X]**	Determines the encryption type used in the main mode negotiation for up to four proposals. X stands for the proposal number (0 to 3). The valid encryption values are: Not Defined (default) DES-CBC **[1]** Triple DES-CBC **[2]** AES **[3]**
First to Fourth Proposal Authentication Type **[IKEPolicyProposalAuthentication_X]**	Determines the authentication protocol used in the main mode negotiation for up to four proposals. X stands for the proposal number (0 to 3). The valid authentication values are: Not Defined (default) HMAC-SHA1-96)**[2]** HMAC-MD5-96 **[4]**
First to Fourth Proposal DH Group **[IKEPolicyProposalDHGroup_X]**	Determines the length of the key created by the DH protocol for up to four proposals. X stands for the proposal number (0 to 3). The valid DH Group values are: Not Defined (default) DH-786-Bit **[0]** DH-1024-Bit **[1]**

Table 13-1: IKE Table Configuration Parameters (continues on pages 283 to 284)

Parameter Name	Description
Authentication Method **[IkePolicyAuthenticationMethod]**	Determines the authentication method for IKE. The valid authentication method values include: 0 = Pre-shared Key (default) 1 = RSA Signiture **Note 1:** For pre-shared key based authentication, peers participating in an IKE exchange must have a prior (out-of-band) knowledge of the common key (see IKEPolicySharedKey parameter). **Note 2:** For RSA signature based authentication, peers must be loaded with a certificate signed by a common CA. For additional information on certificates, refer to Section 13.2.4 on page 291.
IKE SA LifeTime (sec) **[IKEPolicyLifeInSec]**	Determines the time (in seconds) the SA negotiated in the first IKE session (main mode) is valid. After the time expires, the SA is re-negotiated. The default value is 28800 (8 hours).
IKE SA LifeTime (KB) **[IKEPolicyLifeInKB]**	Determines the lifetime (in kilobytes) the SA negotiated in the first IKE session (main mode) is valid. After this size is reached, the SA is re-negotiated. The default value is 0 (this parameter is ignored).
The lifetime parameters (IKEPolicyLifeInSec and IKEPolicyLifeInKB) determine the duration the SA created in the main mode phase is valid. When the lifetime of the SA expires, it is automatically renewed by performing the IKE first phase negotiations. To refrain from a situation where the SA expires, a new SA is being negotiated while the old one is still valid. As soon as the new SA is created, it replaces the old one. This procedure occurs whenever an SA is about to expire.	

If no IKE methods are defined (Encryption / Authentication / DH Group), the default settings (shown in Table 13-2 below) are applied.

Table 13-2: Default IKE First Phase Proposals

	Encryption	Authentication	DH Group
Proposal 0	3DES	SHA1	1024
Proposal 1	3DES	MD5	1024
Proposal 2	3DES	SHA1	786
Proposal 3	3DES	MD5	786

➢ To configure the IKE table using the *ini* file:

The IKE parameters are configured using *ini* file tables (described in Section 11.5 on page 253). Each line in the table refers to a different IKE peer.

The Format line (IKE_DB_INDEX in the example below) specifies the order in which the actual data lines are written. The order of the parameters is irrelevant. Parameters are not mandatory unless stated otherwise. To support more than one Encryption / Authentication / DH Group proposals, for *each* proposal specify the relevant parameters in the Format line. Note that the proposal list must be contiguous.

Figure 13-2: Example of an IKE Table

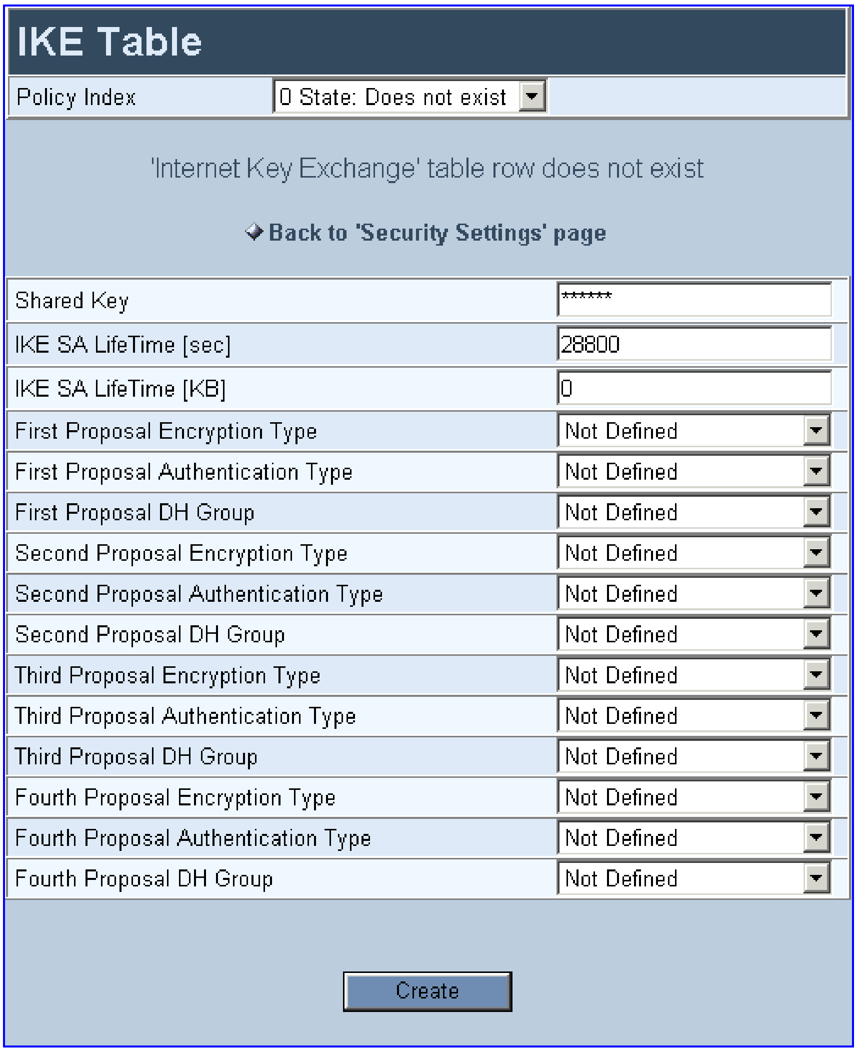
```
[IPSec_IKEDB_Table]

Format IKE_DB_INDEX = IKEPolicySharedKey, IKEPolicyProposalEncryption_0,
IKEPolicypRoposalAuthentication_0, IKEPolicyProposalDHGroup_0,
IKEPolicyProposalEncryption_1, IKEPolicypRoposalAuthentication_1,
IKEPolicyProposalDHGroup_1, IKEPolicyLifeInSec;

IPSEC_IKEDB_TABLE 0 = 123456789, 1, 2, 0, 2, 2, 1, 28800;

[\IPSEC_IKEDB_TABLE]
```

In the example, a single IKE peer is configured. Its pre-shared key is 123456789. Two security proposals are configured: DES/SHA1/786DH and 3DES/SHA1/1024DH.

➢ To configure the IKE table using the Embedded Web Server, take these 6 steps:

1. Access the Embedded Web Server (refer to Section 5.3 on page 58).
2. Open the 'IKE Table' screen (**Advanced Configuration** menu > **Security Settings** > **IKE Table** option); the 'IKE Table' screen is displayed.

Figure 13-3: IKE Table Screen



3. In the 'Policy Index' drop-down list, select the peer you want to edit (up to 20 peers can be configured).
4. Configure the IKE parameters according to Table 13-1 on page 283.
5. Click the button **Create**; a row is create in the IKE table
6. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

To delete a peer from the IKE table, select it in the 'Policy Index' drop-down list, click the button **Delete** and click **OK** at the prompt.

13.1.3.2 IPSec Configuration

The parameters described in Table 13-3 below are used to configure the SPD table. A different set of parameters can be configured for each of the 20 available IP destinations.

Table 13-3: SPD Table Configuration Parameters (continues on pages 286 to 286)

<table>
<tr><th>Parameter Name</th><th colspan="2">Description</th></tr>
<tr><td>Remote IP Address
[IPSecPolicyRemoteIPAddress]</td><td>Defines the destination IP address (or a FQDN) the IPSec mechanism is applied to.
This parameter is mandatory.
Note: When a FQDN is used, a DNS server must be configured (DNSPriServerIP).</td><td rowspan="5">IPSec is applied to outgoing packets whose IP address, destination port, source port and protocol type match the values defined for these four parameters.</td></tr>
<tr><td>Local IP Address Type
[IPSecPolicyLocalIPAddressType]</td><td>Determines the local interface to which the encryption is applied (applicable to multiple IPs and VLANs).
0 = OAM interface (default).
1 = Control interface.</td></tr>
<tr><td>Source Port
[IPSecPolicySrcPort]</td><td>Defines the source port the IPSec mechanism is applied to.
The default value is 0 (any port).</td></tr>
<tr><td>Destination Port
[IPSecPolicyDstPort]</td><td>Defines the destination port the IPSec mechanism is applied to.
The default value is 0 (any port).</td></tr>
<tr><td>Protocol
[IPSecPolicyProtocol]</td><td>Defines the protocol type the IPSec mechanism is applied to.
0 = Any protocol (default).
17 = UDP.
6 = TCP.
Or any other protocol type defined by IANA (Internet Assigned Numbers Authority).</td></tr>
<tr><td>Related Key Exchange Method Index
[IPsecPolicyKeyExchangeMethodIndex]</td><td colspan="2">Determines the index for the corresponding IKE entry. Note that several policies can be associated with a single IKE entry.
The valid range is 0 to 19. The default value is 0.</td></tr>
<tr><td colspan="3">**IKE Second Phase Parameters (Quick Mode)**</td></tr>
<tr><td>SA Lifetime (sec)
[IPsecPolicyLifeInSec]</td><td colspan="2">Determines the time (in seconds) the SA negotiated in the second IKE session (quick mode) is valid. After the time expires, the SA is re-negotiated.
The default value is 28800 (8 hours).</td></tr>
<tr><td>SA Lifetime (KB)
[IPSecPolicyLifeInKB]</td><td colspan="2">Determines the lifetime (in kilobytes) the SA negotiated in the second IKE session (quick mode) is valid. After this size is reached, the SA is re-negotiated.
The default value is 0 (this parameter is ignored).</td></tr>
<tr><td colspan="3">The lifetime parameters (IPsecPolicyLifeInSec and IPSecPolicyLifeInKB) determine the duration of which an SA is valid. When the lifetime of the SA expires, it is automatically renewed by performing the IKE second phase negotiations. To refrain from a situation where the SA expires, a new SA is being negotiated while the old one is still valid. As soon as the new SA is created, it replaces the old one. This procedure occurs whenever an SA is about to expire.</td></tr>
<tr><td>First to Fourth Proposal Encryption Type
[IPSecPolicyProposalEncryption_X]</td><td colspan="2">Determines the encryption type used in the quick mode negotiation for up to four proposals.
X stands for the proposal number (0 to 3).
The valid encryption values are:
Not Defined (default)
None **[0]** = No encryption
DES-CBC **[1]**
Triple DES-CBC **[2]**
AES **[3]**</td></tr>
<tr><td>First to Fourth Proposal Authentication Type
[IPSecPolicyProposalAuthentication_X]</td><td colspan="2">Determines the authentication protocol used in the quick mode negotiation for up to four proposals.
X stands for the proposal number (0 to 3).
The valid authentication values are:
Not Defined (default)
HMAC-SHA-1-96**[2]**
HMAC-MD5-96 **[4]**</td></tr>
</table>

If no IPSec methods are defined (Encryption / Authentication), the default settings (shown in Table 13-4 below) are applied.

Table 13-4: Default IKE Second Phase Proposals

	Encryption	Authentication
Proposal 0	3DES	SHA1
Proposal 1	3DES	MD5
Proposal 2	DES	SHA1
Proposal 3	DES	MD5

➢ To configure the SPD table using the *ini* file:

SPD table is configured using *ini* file tables (described in Section 11.5 on page 253). Each line in the table refers to a different IP destination.

The Format line (SPD_INDEX in the example below) specifies the order in which the actual data lines are written. The order of the parameters is irrelevant. Parameters are not mandatory unless stated otherwise. To support more than one Encryption / Authentication proposals, for *each* proposal specify the relevant parameters in the Format line. Note that the proposal list must be contiguous.

Figure 13-4: Example of an SPD Table

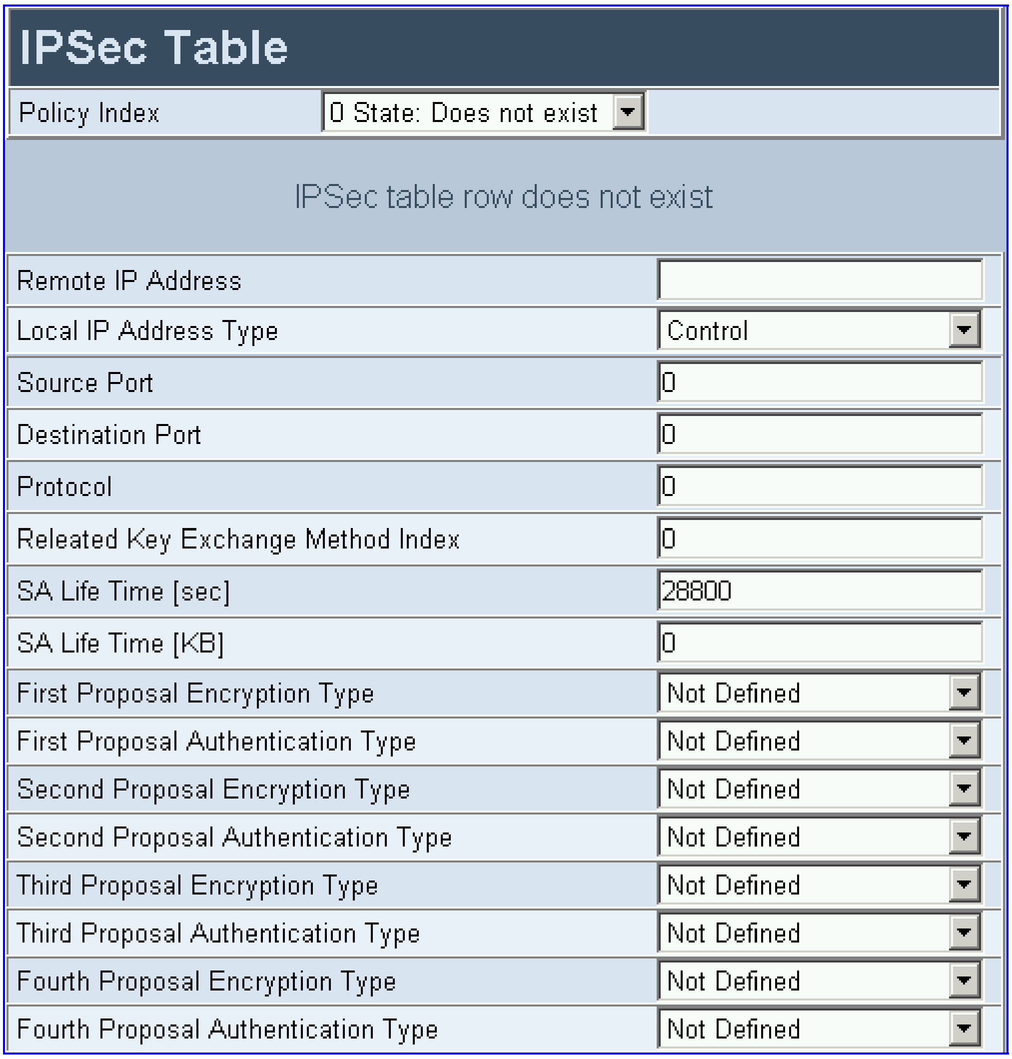
```
[ IPSEC_SPD_TABLE ]

Format SPD_INDEX = IPSecPolicyRemoteIPAddress, IpsecPolicySrcPort,
IPSecPolicyDStPort,IPSecPolicyProtocol, IPSecPolicyLifeInSec,
IPSecPolicyProposalEncryption_0, IPSecPolicyProposalAuthentication_0,
IPSecPolicyProposalEncryption_1, IPSecPolicyProposalAuthentication_1,
IPSecPolicyKeyExchangeMethodIndex, IPSecPolicyLocalIPAddressType;

IPSEC_SPD_TABLE 0 = 10.11.2.21, 0, 0, 17, 900, 1,2, 2,2 ,1, 0;

[ \IPSEC_SPD_TABLE ]
```

In the SPD example above, all packets designated to IP address 10.11.2.21 that originates from the OAM interface (regardless to their destination and source ports) and whose protocol is UDP are encrypted, the SPD also defines an SA lifetime of 900 seconds and two security proposals: DES/SHA1 and 3DES/SHA1.

➢ To configure the SPD table using the Embedded Web Server, take these 6 steps:

1. Access the Embedded Web Server (refer to Section 5.3 on page 58).
2. Open the 'IPSec Table' screen (**Advanced Configuration** menu > **Security Settings** > **IPSec Table** option); the 'IPSec Table' screen is displayed.

Figure 13-5: IPSec Table Screen



3. In the 'Policy Index' drop-down list, select the rule you want to edit (up to 20 rules can be configured).
4. Configure the SPD parameters according to Table 13-3 on page 286.
5. Click the button **Create**; a row is create in the SPD table
6. To save the changes so they are available after a power fail, refer to Section 5.9.2 on page 124.

To delete a peer from the SPD table, select it in the 'Policy Index' drop-down list, click the button **Delete** and click **OK** at the prompt.

13.1.3.3 IPSec and IKE Configuration Table's Confidentiality

Since the pre-shared key parameter of the IKE table must remain undisclosed, measures are taken by the *ini* file, Embedded Web Server and SNMP agent to maintain this parameter's confidentiality. On the Embedded Web Server a list of asterisks is displayed instead of the pre-shared key. On SNMP, the pre-shared key parameter is a write-only parameter and cannot be read. In the *ini* file, the following measures to assure the secrecy of the IPSec and IKE tables are taken:

- Hidden IPSec and IKE tables **-** When uploading the *ini* file from the gateway the IPSec and IKE tables are not available. Instead, the notifications (shown in Figure 13-6) are displayed.

Figure 13-6: Example of an *ini* File Notification of Missing Tables

```
;
;  *** TABLE IPSEC_IKEDB_TABLE ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts
;

;
;  *** TABLE IPSEC_SPD_TABLE ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts
;
```

- Preserving the values of the parameters in the IPSec and IKE tables from one *ini* file loading to the next – The values configured for the parameters in the IPSec tables in the *ini* file are preserved from one loading to another. If a newly loaded *ini* file doesn't define IPSec tables, the previously loaded tables remain valid. To invalidate a previously loaded *ini* file's IPSec tables, load a new *ini* file with an empty IPSec table (shown below).

Figure 13-7: Empty IPSec / IKE Tables

```
[IPSec_IKEDB_Table]
[\IPSec_IKEDB_Table]

[IPSEC_SPD_TABLE]
[\IPSEC_SPD_TABLE]
```

13.2 SSL/TLS

SSL, also known as TLS, is the method used to secure the gateway's SIP Signaling connections, Embedded Web Server and Telnet server. The SSL protocol provides confidentiality, integrity and authenticity between two communicating applications over TCP/IP.

Specifications for the SSL/TLS implementation:

- Supports transports: SSL 2.0, SSL 3.0, TLS 1.0
- Supports ciphers: DES, RC4 compatible
- Authentication: X.509 certificates; CRLs are not supported

13.2.1 SIP Over TLS (SIPS)

The gateway uses TLS over TCP to encrypt SIP transport and (optionally) to authenticate it. To enable TLS on the gateway, set the selected transport type to TLS (SIPTransportType = 2). In this mode the gateway initiates a TLS connection only for the next network hop. To enable TLS all the way to the destination (over multiple hops) set EnableSIPS to 1. When a TLS connection with the gateway is initiated, the gateway also responds using TLS regardless of the configured SIP transport type (in this case, the parameter EnableSIPS is also ignored).

TLS and SIPS use the Certificate Exchange process described in Sections 13.2.4 and 13.2.5. To change the port number used for SIPS transport (by default 5061), use the parameter, TLSLocalSIPPort.

When SIPS is used, it is sometimes required to use two-way authentication. When acting as the TLS server (in a specific connection) it is possible to demand the authentication of the client's certificate. To enable two-way authentication on the gateway, set the *ini* file parameter, SIPSRequireClientCertificate = 1. For information on installing a client certificate, refer to Section 13.2.5 on page 293.

13.2.2 Embedded Web Server Configuration

For additional security, you can configure the Embedded Web Server to accept only secured (HTTPS) connections by changing the parameter HTTPSOnly to 1 (described in Table 6-3 on page 143).

You can also change the port number used for the secured Web server (by default 443) by changing the *ini* file parameter, HTTPSPort (described in Table 6-3 on page 143).

13.2.2.1 Using the Secured Embedded Web Server

➢ **To use the secured Embedded Web Server, take these 3 steps:**

1. Access the gateway using the following URL:
https://[*host name] or [IP address]*
Depending on the browser's configuration, a security warning dialog may be displayed. The reason for the warning is that the gateway initial certificate is not trusted by your PC. The browser may allow you to install the certificate, thus skipping the warning dialog the next time you connect to the gateway.
2. If you are using Internet Explorer, click **View Certificate** and then **Install Certificate**.
3. The browser also warns you if the host name used in the URL is not identical to the one listed in the certificate. To solve this, add the IP address and host name (ACL_nnnnnn where nnnnnn is the serial number of the gateway) to your hosts file, located at /etc/hosts on UNIX or C:\Windows\System32\Drivers\ETC\hosts on Windows; then use the host name in the URL (e.g., https://ACL_280152).

The figure below is an example of a host file:

Figure 13-8: Example of a Host File

```
# This is a sample HOSTS file used by Microsoft TCP/IP for Windows.
# Location: C:\WINDOWS\SYSTEM32\DRIVERS\ETC\hosts
#
127.0.0.1     localhost
10.31.4.47    ACL_280152
```

13.2.3 Secured Telnet

To enable the embedded Telnet server on the gateway, set the parameter TelnetServerEnable (described in Table 6-3 on page 143) to 1 (standard mode) or 2 (SSL mode); no information is transmitted in the clear when SSL mode is used.

If the Telnet server is set to SSL mode, a special Telnet client is required on your PC to connect to the Telnet interface over a secured connection; examples include C-Kermit for UNIX, Kermit-95 for Windows, and AudioCodes' acSSLTelnet utility for Windows (that requires prior installation of the free OpenSSL toolkit). Contact AudioCodes to obtain the acSSLTelnet utility.

13.2.4 Server Certificate Replacement

The gateway is supplied with a working SSL configuration consisting of a unique self-signed server certificate. When the gateway is upgraded to a later firmware version, a unique self-signed server certificate is created. If an organizational Public Key Infrastructure (PKI) is used, you may wish to replace this certificate with one provided by your security administrator.

➢ To replace the gateway's self-signed certificate, take these 9 steps:

1. Your network administrator should allocate a unique DNS name for the gateway (e.g., dns_name.corp.customer.com). This name is used to access the device, and should therefore be listed in the server certificate.
2. Open the 'Certificate' screen (**Advanced Configuration** menu > **Security Settings** submenu > **Certificates** option); the 'Certificates' screen is displayed (Figure 13-9).

Figure 13-9: Certificate Signing Request Screen

3. In the 'Subject Name' field, enter the DNS name and click **Generate CSR**. A textual certificate signing request, that contains the SSL device identifier, is displayed.
4. Copy this text and send it to your security provider; the security provider (also known as Certification Authority or CA) signs this request and send you a server certificate for the device.
5. Save the certificate in a file (e.g., cert.txt). Ensure the file is a plain-text file with the 'BEGIN CERTIFICATE' header. The figure below is an example of a Base64-Encoded X.509 Certificate.

Figure 13-10: Example of a Base64-Encoded X.509 Certificate


```
-----BEGIN CERTIFICATE-----
MIIDkzCCAnugAwIBAgIEAgAAADANBgkqhkiG9w0BAQQFADA/MQswCQYDVQQGEwJG
UjETMBEGA1UEChMKQ2VydGlwb3N0ZTEbMBkGA1UEAxMSQ2VydGlwb3N0ZSBTZXJ2
ZXVyMB4XDTk4MDYyNDA4MDAwMFoXDTE4MDYyNDA4MDAwMFowPzELMAkGA1UEBhMC
RlIxEzARBgNVBAoTCkNlcnRpcG9zdGUxGzAZBgNVBAMTEkNlcnRpcG9zdGUgU2Vy
dmV1cjCCASEwDQYJKoZIhvcNAQEBBQADggEOADCCAQkCggEAPqd4MziR4spWldGR
x8bQrhZkonWnNm`+Yhb7+4Q67ecf1janH7GcN/SXsfx7jJpreWULf7v7Cvpr4R7qI
JcmdHIntmf7JPM5n6cDBv17uSW63er7NkVnMFHwK1QaGFLMybFkzaeGrvFm4k3lR
efiXDmuOe+FhJgHYezYHf44LvPRPwhSrzi9+Aq3o8pWDguJuZDIUP1F1jMa+LPwv
REXfFcUW+w==
-----END CERTIFICATE-----
```


6. Before continuing, set the parameter HTTPSOnly = 0 to ensure you have a method of accessing the device in case the new certificate doesn't work. Restore the previous setting after testing the configuration.
7. In the 'Certificate' screen (Figure 13-9) locate the server certificate loading section.

8. Click **Browse**, navigate to the *cert.txt* file, and then click **Send File**.
9. When the operation is completed, save the configuration (Section 5.9.2 on page 124) and restart the gateway; the Embedded Web Server uses the provided certificate.

Notes:

- The certificate replacement process can be repeated when necessary (e.g., the new certificate expires).
- It is possible to use the IP address of the gateway (e.g., 10.3.3.1) instead of a qualified DNS name in the Subject Name. This practice is not recommended since the IP address is subject to changes and may not uniquely identify the device.
- The server certificate can also be loaded via *ini* file using the parameter 'HTTPSCertFileName'.

13.2.5 Client Certificates

By default, Web servers using SSL provide one-way authentication. The client is certain that the information provided by the Web server is authentic. When an organizational PKI is used, two-way authentication may be desired: both client and server should be authenticated using X.509 certificates. This is achieved by installing a client certificate on the managing PC, and loading the same certificate (in base64-encoded X.509 format) to the gateway's Trusted Root Certificate Store. The Trusted Root Certificate file should contain both the certificate of the authorized user and the certificate of the CA.

Since X.509 certificates have an expiration date and time, the gateway must be configured to use NTP (Section 9.8 on page 236) to obtain the current date and time. Without a correct date and time, client certificates cannot work.

➢ To install a client certificate, take these 6 steps:

1. Before continuing, set HTTPSOnly = 0 to ensure you have a method of accessing the device in case the client certificate doesn't work. Restore the previous setting after testing the configuration.
2. Open the 'Certificates' screen (**Advanced Configuration** menu > **Security Settings** submenu > **Certificates** option); the 'Certificates' screen is displayed (Figure 13-9).
3. To load the Trusted Root Certificate file locate the trusted root certificate loading section.
4. Click **Browse**, navigate to the file, and then click **Send File**.
5. When the operation is completed, set the *ini* file parameter, HTTPSRequireClientCertificates = 1.
6. Save the configuration (Section 5.9.2 on page 124) and restart the gateway.

When a user connects to the secure Web server:

- If the user has a client certificate from a CA listed in the Trusted Root Certificate file, the connection is accepted and the user is prompted for the system password.
- If both the CA certificate and the client certificate appear in the Trusted Root Certificate file, the user is not prompted for a password (thus providing a single-sign-on experience - the authentication is performed using the X.509 digital signature).
- If the user doesn't have a client certificate from a listed CA, or doesn't have a client certificate at all, the connection is rejected.

Notes:

- The process of installing a client certificate on your PC is beyond the scope of this document. For more information, refer to your Web browser or operating system documentation, and/or consult your security administrator.
- The root certificate can also be loaded via *ini* file using the parameter 'HTTPSRootFileName'.

13.3 SRTP

The gateway supports Secured RTP (SRTP) according to RFC 3711. SRTP is used to encrypt RTP and RTCP transport since it is best-suited for protecting VoIP traffic.

SRTP requires a Key Exchange mechanism that is performed according to <draft-ietf-mmusic-sdescriptions-12>. The Key Exchange is executed by adding a 'Crypto' attribute to the SDP. This attribute is used (by both sides) to declare the various supported cipher suites and to attach the encryption key to use. If negotiation of the encryption data is successful, the call is established.

Use the parameter MediaSecurityBehaviour (described in Section 6.9 on page 145) to select the gateway's mode of operation: Must or Prefer. These modes determine the behavior of the gateway if negotiation of the cipher suite fails.

- Must = the call is terminated. Incoming calls that don't include encryption information are rejected.
- Prefer = an unencrypted call is established. Incoming calls that don't include encryption information are accepted.

To enable SRTP set the parameter EnableMediaSecurity to 1 (described in Section 6.9 on page 145).

Notes:

- When SRTP is used the channel capacity is reduced (refer to the parameter EnableMediaSecurity).
- The gateway only supports the AES 128 in CM mode cipher suite.

Figure 13-11: Example of crypto Attributes Usage

```
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:PsKoMpHlCg+b5X0YLuSvNrImEh/dAe
a=crypto:2 AES_CM_128_HMAC_SHA1_32 inline:IsPtLoGkBf9a+c6XVzRuMqHlDnEiAd
```

13.4 RADIUS Login Authentication

Users can enhance the security and capabilities of logging to the gateway's Web and Telnet embedded servers by using a Remote Authentication Dial-In User Service (RADIUS) to store numerous usernames, passwords and access level attributes (Web only), allowing multiple user management on a centralized platform. RADIUS (RFC 2865) is a standard authentication protocol that defines a method for contacting a predefined server and verifying a given name and password pair against a remote database, in a secure manner.

When accessing the Web and Telnet servers, users must provide a valid username and password. When RADIUS authentication isn't used, the username and password are authenticated with the Embedded Web Server's usernames and passwords of the primary or secondary accounts (refer to Section 5.2.1 on page 56) or with the Telnet server's username and password stored internally in the gateway's memory. When RADIUS authentication is used, the gateway doesn't store the username and password but simply forwards them to the pre-configured RADIUS server for authentication (acceptance or rejection). The internal Web / Telnet passwords can be used as a fallback mechanism in case the RADIUS server doesn't respond (configured by the parameter BehaviorUponRadiusTimeout). Note that when RADIUS authentication is performed, the Web / Telnet servers are blocked until a response is received (with a timeout of 5 seconds).

RADIUS authentication requires HTTP basic authentication, meaning the username and password are transmitted in clear text over the network. Therefore, users are recommended to set the parameter 'HttpsOnly = 1' to force the use of HTTPS, since the transport is encrypted.

13.4.1 Setting Up a RADIUS Server

The following examples refer to FreeRADIUS, a free RADIUS server that can be downloaded from www.freeradius.org. Follow the directions on that site for information on installing and configuring the server. If you use a RADIUS server from a different vendor, refer to its appropriate documentation.

➢ To set up a RADIUS server, take these 5 steps:

1. Define the gateway as an authorized client of the RADIUS server, with a predefined 'shared secret' (a password used to secure communication) and a vendor ID. The figure below displays an example of the file clients.conf (FreeRADIUS client configuration).

Figure 13-12: Example of the File clients.conf (FreeRADIUS Client Configuration)

```
#
# clients.conf - client configuration directives
#
client 10.31.4.47 {
        secret          = FutureRADIUS
        shortname       = tp1610_master_tpm
}
```

2. If access levels are required, set up a VSA dictionary for the RADIUS server and select an attribute ID that represents each user's access level. The following example shows a dictionary file for FreeRADIUS that defines the attribute 'ACL-Auth-Level' with ID=35.

Figure 13-13: Example of a Dictionary File for FreeRADIUS (FreeRADIUS Client Configuration)

```
#
# AudioCodes VSA dictionary
#
VENDOR AudioCodes 5003
ATTRIBUTE ACL-Auth-Level 35 integer AudioCodes
VALUE ACL-Auth-Level ACL-Auth-UserLevel 50
VALUE ACL-Auth-Level ACL-Auth-AdminLevel 100
VALUE ACL-Auth-Level ACL-Auth-SecurityAdminLevel 200
```

3. In the RADIUS server, define the list of users authorized to use the gateway, using one of the password authentication methods supported by the server implementation. The following example shows a user configuration file for FreeRADIUS using a plain-text password.

Figure 13-14: Example of a User Configuration File for FreeRADIUS Using a Plain-Text Password

```
# users - local user configuration database

john     Auth-Type := Local, User-Password == "qwerty"
         Service-Type = Login-User,
         ACL-Auth-Level = ACL-Auth-SecurityAdminLevel

larry    Auth-Type := Local, User-Password == "123456"
         Service-Type = Login-User,
         ACL-Auth-Level = ACL-Auth-UserLevel
```

4. Record and retain the IP address, port number, 'shared secret', vendor ID and VSA access level identifier (if access levels are used) used by the RADIUS server.
5. Configure the gateway's relevant parameters according to Section 13.4.2 below.

13.4.2 Configuring RADIUS Support

For information on the RADIUS parameters, refer to Table 6-5 on page 147.

➢ **To configure RADIUS support on the gateway via the Embedded Web Server, take these 13 steps:**

1. Access the Embedded Web Server (Section 5.3 on page 58).
2. Open the 'General Security Settings' screen (**Advanced Configuration** menu > **Security Settings** > **General Security Settings** option); the 'General Security Settings' screen is displayed.
3. Under section 'General RADIUS Settings', in the field 'Enable RADIUS Access Control', select 'Enable'; the RADIUS application is enabled.
4. In the field 'Use RADIUS for Web / Telnet Login', select 'Enable'; RADIUS authentication is enabled for Web and Telnet login.
5. Enter the RADIUS server IP address, port number and shared secret in the relevant fields.
6. Under section 'RADIUS Authentication Settings', in the field 'Device Behavior Upon RADIUS Timeout', select the gateway's operation if a response isn't received from the RADIUS server after the 5 seconds timeout expires:
   - Deny Access: gateway denies access to the Web and Telnet embedded servers.
   - Verify Access Locally: gateway checks the local username and password.

7. In the field 'Local RADIUS Password Cache Timeout', enter a time (in seconds); when this time expires, the username and password verified by the RADIUS server becomes invalid and a username and password must be re-validated with the RADIUS server.
8. In the field 'Local RADIUS Password Cache Mode', select the gateway's mode of operation regarding the above-mentioned 'Local RADIUS Password Cache Timer' option:
   - Reset Timer Upon Access: upon each access to a Web screen, the timer resets (reverts to the initial value configured in the previous step).
   - Absolute Expiry Timer: when you access a Web screen, the timer doesn't reset but rather continues decreasing.
9. In the field 'RADIUS VSA Vendor ID', enter the vendor ID you configured in the RADIUS server:
10. When using the Web access-level mechanism, perform one of the following options:
    - When RADIUS responses include the access level attribute:
      In the field 'RADIUS VSA Access Level Attribute', enter the code that indicates the access level attribute in the Vendor Specific Attributes (VSA) section of the received RADIUS packet.
    - When RADIUS responses don't include the access level attribute:
      In the field 'Default Access Level', enter the default access level that is applied to all users authenticated by the RADIUS server.
11. In the field 'Require Secured Web Connection (HTTPS)', select 'HTTPS only'. It is important you use HTTPS (secure Web server) when connecting to the gateway over an open network, since the password is transmitted in clear text. Similarly, for Telnet, use SSL 'TelnetServerEnable = 2 (refer to Section 13.2.3 on page 291).
12. To save the changes, refer to Section 5.9.2 on page 124.
13. Reset the gateway (Section 5.9.3 on page 125).

After reset, when accessing the Web or Telnet servers, use the username and password you configured in the RADIUS database. The local system password is still active and can be used when the RADIUS server is down.

➢ To configure RADIUS support on the gateway using the *ini* file:

- Add the following parameters to the *ini* file. For information on modifying the *ini* file, refer to Section 6.2 on page 127.
  - EnableRADIUS = 1
  - WebRADIUSLogin = 1
  - RADIUSAuthServerIP = *IP address of RADIUS server*
  - RADIUSAuthPort = *port number of RADIUS server, usually 1812*
  - SharedSecret = *your shared secret*'
  - HTTPSOnly = 1
  - BehaviorUponRadiusTimeout = 1
  - RadiusLocalCacheMode = 1
  - RadiusLocalCacheTimeout = 300
  - RadiusVSAVendorID = *your vendor's ID*
  - RadiusVSAAccessAttribute = *code that indicates the access level attribute*
  - DefaultAccessLevel = *default access level (0 to 200)*

13.5 Internal Firewall

The gateway accommodates an internal access list facility, allowing the security administrator to define network traffic filtering rules. The access list provides the following features:

- Block traffic from known malicious sources
- Only allow traffic from known friendly sources, and block all others
- Mix allowed and blocked network sources
- Limit traffic to a predefined rate (blocking the excess)
- Limit traffic to specific protocols, and specific port ranges on the device

The access list consists of a table with up to 50 ordered lines. For each packet received on the network interface, the table is scanned from the top until a matching rule is found (or the table end is reached). This rule can either block the packet or allow it; however it is important to note that subsequent rules aren't scanned. If the table end is reached without a match, the packet is accepted.

Each rule is composed of the following fields (described in Table 6-1 on page 130):

- IP address (or DNS name) of source network
- IP network mask
- Destination UDP/TCP ports (on this device)
- Protocol type
- Maximum packet size, byte rate per second, and allowed data burst
- Action upon match (allow or block)

Figure 13-15 shows an example of an access list definition via *ini* file:

Figure 13-15: Example of an Access List Definition via *ini* File

```
[ ACCESSLIST ]
FORMAT AccessList_Index = AccessList_Source_IP, AccessList_Net_Mask,
AccessList_Start_Port, AccessList_End_Port, AccessList_Protocol,
AccessList_Packet_Size, AccessList_Byte_Rate, AccessList_Byte_Burst,
AccessList_Allow_Type;

AccessList 10 = mgmt.customer.com, 255.255.255.255, 0, 80, tcp, 0, 0, 0, allow ;
AccessList 15 = 192.0.0.0, 255.0.0.0, 0, 65535, any, 0, 40000, 50000, block ;
AccessList 20 = 10.31.4.0, 255.255.255.0, 4000, 9000, any, 0, 0, 0, block ;
AccessList 22 = 10.4.0.0, 255.255.0.0, 4000, 9000, any, 0, 0, 0, block ;
[ \ACCESSLIST ]
```

Explanation of the example access list:

- Rule #10: traffic from the host 'mgmt.customer.com' destined to TCP ports 0 to 80, is always allowed.
- Rule #15: traffic from the 192.xxx.yyy.zzz subnet, is limited to a rate of 40 Kbytes per second (with an allowed burst of 50 Kbytes). Note that the rate is specified in bytes, not bits, per second; a rate of 40000 bytes per second, nominally corresponds to 320 kbps.
- Rule #20: traffic from the subnet 10.31.4.xxx destined to ports 4000 to 9000 is always blocked, regardless of protocol.
- Rule #22: traffic from the subnet 10.4.xxx.yyy destined to ports 4000 to 9000 is always blocked, regardless of protocol.
- All other traffic is allowed.

More complex rules may be defined, relying on the 'single-match' process described above.

Figure 13-16 shows an advanced example of an access list definition via *ini* file:

Figure 13-16: Advanced Example of an Access List Definition via *ini* File

```
[ ACCESSLIST ]
FORMAT AccessList_Index = AccessList_Source_IP, AccessList_Net_Mask,
AccessList_Start_Port, AccessList_End_Port, AccessList_Protocol,
AccessList_Packet_Size, AccessList_Byte_Rate, AccessList_Byte_Burst,
AccessList_Allow_Type;

AccessList 10 = 10.0.0.0, 255.0.0.0, 0, 65535, any, 0, 40000, 50000, allow ;
AccessList 15 = 10.31.4.0, 255.255.255.0, 4000, 9000, any, 0, 0, 0, allow ;
AccessList 20 = 0.0.0.0, 0.0.0.0, 0, 65535, any, 0, 0, 0, block;
 [ \ACCESSLIST ]
```

Explanation of the example access list:

This access list consists of three rules:

- Rule #10: traffic from the subnet 10.xxx.yyy.zzz is allowed if the traffic rate does not exceed 40 KB/s.
- Rule #15: if a packet didn't match rule #10, that is, the excess traffic is over 40 KB/s, and coming from the subnet 10.31.4.xxx to ports 4000 to 9000, then it is allowed.
- Rule #20: all other traffic (which didn't match the previous rules), is blocked.

The internal firewall can also be configured via the Embedded Web Server (refer to Section 5.6.8.3 on page 101).

13.6 Network Port Usage

The following table lists the default TCP/UDP network port numbers used by the gateway. Where relevant, the table lists the *ini* file parameters that control the port usage and provide source IP address filtering capabilities.

Table 13-5: Default TCP/UDP Network Port Numbers

Port Number	Peer Port	Application	Notes
2	2	Debugging interface	Always ignored
23	-	Telnet	Disabled by default (TelnetServerEnable). Configurable (TelnetServerPort), access controlled by WebAccessList
68	67	DHCP	Active only if DHCPEnable = 1
80	-	Web server (HTTP)	Configurable (HTTPPort), can be disabled (DisableWebTask or HTTPSOnly). Access controlled by WebAccessList
161	-	SNMP GET/SET	Configurable (SNMPPort), can be disabled (DisableSNMP). Access controlled by SNMPTrustedMGR
443	-	Web server (HTTPS)	Configurable (HTTPSPort), can be disabled (DisableWebTask). Access controlled by WebAccessList
500	-	IPSec IKE	Can be disabled (EnableIPSec) Not supported in the current version.
6000, 6010 and up	-	RTP traffic	Base port number configurable (BaseUDPPort), fixed increments of 10. The number of ports used depends on the channel capacity of the device.
6001, 6011 and up	-	RTCP traffic	Always adjacent to the RTP port number
6002, 6012 and up	-	T.38 traffic	Always adjacent to the RTCP port number
5060	5060	SIP	Configurable (LocalSIPPort [UDP], TCPLocalSIPPort [TCP]).
5061	5061	SIP over TLS (SIPS)	Configurable (TLSLocalSIPPort)
(random) > 32767	514	Syslog	Disabled by default (EnableSyslog).
(random) > 32767	-	Syslog ICMP	Disabled by default (EnableSyslog).
(random) > 32767	-	ARP listener	
(random) > 32767	162	SNMP Traps	Can be disabled (DisableSNMP)
(random) > 32767	-	DNS client	

13.7 Recommended Practices

To improve network security, the following guidelines are recommended when configuring the gateway:

- Set the password of the primary web user account (refer to Section 5.6.8.1 on page 98) to a unique, hard-to-hack string. Do not use the same password for several devices as a single compromise may lead to others. Keep this password safe at all times and change it frequently.
- If possible, use a RADIUS server for authentication. RADIUS allows you to set different passwords for different users of the gateway, with centralized management of the password database. Both Web and Telnet interfaces support RADIUS authentication (refer to Section 13.3 on page 294).
- If the number of users that access the Web and Telnet interfaces is limited, you can use the ‘Web and Telnet Access List’ to define up to ten IP addresses that are permitted to access these interfaces. Access from an undefined IP address is denied (refer to Section 5.6.8.2 on page 100).
- Use IPSec to secure traffic to all management and control hosts. Since IPSec encrypts all traffic, hackers cannot capture sensitive data transmitted on the network, and malicious intrusions are severely limited.
- Use HTTPS when accessing the Web interface. Set HTTPSOnly to 1 to allow only HTTPS traffic (and block port 80). If you don't need the Web interface, disable the Web server (DisableWebTask).
- If you use Telnet, do not use the default port (23). Use SSL mode to protect Telnet traffic from network sniffing.
- If you use SNMP, do not leave the community strings at their default values as they can be easily guessed by hackers (refer to Section 15.7.1 on page 316).
- Use a firewall to protect your VoIP network from external attacks. Network robustness may be compromised if the network is exposed to Denial of Service (DoS) attacks. DoS attacks are mitigated by Stateful firewalls. Do not allow unauthorized traffic to reach the gateway.

13.8 Legal Notice

By default, the gateway supports export-grade (40-bit and 56-bit) encryption due to US government restrictions on the export of security technologies. To enable 128-bit and 256-bit encryption on your device, contact your AudioCodes representative.

This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit (www.openssl.org)

This product includes cryptographic software written by Eric Young' (eay@cryptsoft.com).

Reader's Notes

14 Diagnostics

Several diagnostic tools are provided, enabling you to identify correct functioning of the gateway, or an error condition with a probable cause and a solution or workaround.

- Front panel indicator LEDs on the Mediant 2000. The location and functionality of the front panel LEDs is shown in Section 2.2.1 on page 30.
- Gateway Self-Testing on hardware initialization (refer to Section 14.1 below).
- Syslog Event Notification Messages (refer to Section 14.2).

14.1 Self-Testing

The gateway's self-testing capabilities are used to identify faulty hardware components on startup and during run time.

The gateway features three types of testing modes:

- **Startup tests (Rapid and Enhanced):** These tests have minor impact in real-time. While the Startup tests are executed the regular operation of the gateway is disabled.
    - **Rapid:** The Rapid test is performed every time the gateway starts up. It is executed each time the gateway completes its initialization process. This is a short test phase in which the only error detected and reported is failure in initializing hardware components. If an error is detected, an error message is sent to the Syslog.
    - **Enhanced:** The Enhanced test is performed every time the gateway starts up. The following hardware components are tested:
        - TSA (Time Slot Assigner).
        - PSTN framers (when they are used).
        - Missing DSP's.
        - Conference channels (where they are supported).

  If an error is detected, an error message is sent to the Syslog. Note that when the Detailed test is enabled, errors sent to the Syslog server must be ignored.
- **User-initiated test (Detailed):**The Detailed test is initiated by the user when the gateway is offline (isn't used for regular service).

  Used in addition to the Rapid and Enhanced test modes. The test is performed on startup, when initialization of the gateway is completed and if the parameter EnableDiagnostics is set to 1 or 2. In this mode, the gateway tests its DSPs, RAM and flash memory. When EnableDiagnostics is set to 1, flash is tested thoroughly, when EnableDiagnostics is set to 2, flash is only partially tested. While the Detailed test is running, the Orange LED is lit.
  If an error is detected, an error message is sent to the Syslog.

- **Run-time test (Periodic):** Used for monitoring the gateway during run-time.

  The Periodic Test is performed every hour after startup, even when there is full traffic on the gateway; quality is not degraded. The following hardware components are being tested:

  - TSA.
  - PSTN framers (when they are used).
  - Missing DSP's.
  - Conference channels (where they are supported).
  - If an error is detected, an error message is sent to the Syslog.

Warning: To continue regular operation, the Detailed test must be disabled. Set the parameter EnableDiagnostics to 0 and reset the gateway.

14.2 Syslog Support

Syslog protocol is an event notification protocol that enables a machine to send event notification messages across IP networks to event message collectors- also known as Syslog servers. Syslog protocol is defined in the IETF RFC 3164 standard.

Since each process, application and operating system was written independently, there is little uniformity to Syslog messages. For this reason, no assumption is made on the contents of the messages other than the minimum requirements of its priority.

Syslog uses UDP as its underlying transport layer mechanism. By default, UDP port 514 is assigned to Syslog, but this can be changed (using the SyslogServerPort parameter).

The Syslog message is transmitted as an ASCII (American Standard Code for Information Interchange) message. The message starts with a leading '<' ('less-than' character), followed by a number, which is followed by a '>' ('greater-than' character). This is optionally followed by a single ASCII space.

The number described above is known as the Priority and represents both the Facility and Severity as described below. The Priority number consists of one, two, or three decimal integers.

For example:

```
<37> Oct 11 16:00:15 mymachine su: 'su root' failed for lonvick on
/dev/pts/8
```

Note that when NTP is enabled, a timestamp string [hour:minutes:seconds] is added to all Syslog messages (for information on NTP, refer to Section 9.8 on page 236).

14.2.1 Syslog Servers

Users can use the provided Syslog server (ACSyslog08.exe) or other third-party Syslog servers.

Examples of Syslog servers available as shareware on the Internet:

- Kiwi Enterprises: www.kiwisyslog.com
- The US CMS Server: uscms.fnal.gov/hanlon/uscms_server/
- TriAction Software: www.triaction.nl/Products/SyslogDaemon.asp
- Netal SL4NT 2.1 Syslog Daemon: www.netal.com

A typical Syslog server application enables filtering of the messages according to priority, IP sender address, time, date, etc.

14.2.2 Operation

The Syslog client, embedded in the gateway, sends error reports/events generated by the gateway unit application to a Syslog server, using IP/UDP protocol.

➢ To activate the Syslog client on the gateway, take these 5 steps:

1. Set the parameter ‘EnableSyslog’ to 1 (Table 6-1 on page 130).
2. Use the parameter ‘SyslogServerIP’ to define the IP address of the Syslog server you use (Table 6-1 on page 130).
3. Use the parameter 'SyslogServerPort' to define the port number of the Syslog server (Table 6-1 on page 130).
4. To determine the Syslog logging level use the parameter ‘GWDebugLevel’ (Table 6-2 on page 138).
5. To enable the gateway to send log messages that report certain types of web actions according to a pre-defined filter use the parameter ‘ActivityListToLog’ (described in Table 6-2 on page 138).

Reader's Notes

15 SNMP-Based Management

Simple Network Management Protocol (SNMP) is a standards-based network control protocol for managing elements in a network. The SNMP Manager (usually implemented by a Network Management System (NMS) or an Element Management System (EMS)) connects to an SNMP Agent (embedded on a remote Network Element (NE)) to perform network element Operation, Administration and Maintenance (OAM).

Both the SNMP Manager and the NE refer to the same database to retrieve information or configure parameters. This database is referred to as the Management Information Base (MIB), and is a set of statistical and control values. Apart from the standard MIBs documented in IETF RFCs, SNMP additionally enables the use of private MIBs, containing a non-standard information set (specific functionality provided by the NE).

Directives, issued by the SNMP Manager to an SNMP Agent, consist of the identifiers of SNMP variables (referred to as MIB object identifiers or MIB variables) along with instructions to either get the value for that identifier, or set the identifier to a new value (configuration). The SNMP Agent can also send unsolicited events towards the EMS, called SNMP traps.

The definitions of MIB variables supported by a particular agent are incorporated in descriptor files, written in Abstract Syntax Notation (ASN.1) format, made available to EMS client programs so that they can become aware of MIB variables and their use.

The device contains an embedded SNMP Agent supporting both general network MIBs (such as the IP MIB), VoP-specific MIBs (such as RTP) and AudioCodes' proprietary MIBs (acBoard, acGateway, acAlarm and other MIBs), enabling a deeper probe into the inter-working of the device. All supported MIB files are supplied to customers as part of the release.

15.1 SNMP Standards and Objects

15.1.1 SNMP Message Standard

Four types of SNMP messages are defined:

- **Get:** A request that returns the value of a named object.
- **Get-Next:** A request that returns the next name (and value) of the ‘next’ object supported by a network device given a valid SNMP name.
- **Set:** A request that sets a named object to a specific value.
- **Trap:** A message generated asynchronously by network devices. It is an unsolicited message from an agent to the manager.

Each of these message types fulfills a particular requirement of network managers:

- **Get request:** Specific values can be fetched via the ‘get’ request to determine the performance and state of the device. Typically, many different values and parameters can be determined via SNMP without the overhead associated with logging into the device, or establishing a TCP connection with the device.
- **Get-Next request:** Enables the SNMP standard network managers to ‘walk’ through all SNMP values of a device (via the ‘get-next’ request) to determine all names and values that an operant device supports.
- **Get-Bulk:** Extends the functionality of Get-Next by allowing multiple values to be returned for selected items in the request. This is accomplished by beginning with the first SNMP object to be fetched, fetching the next name with a "get-next", and repeating this operation.

- **Set Request:** The SNMP standard provides a method of effecting an action associated with a device (via the ‘set’ request) to accomplish activities such as disabling interfaces, disconnecting users, clearing registers, etc. This provides a way of configuring and controlling network devices via SNMP.
- **Trap message:** The SNMP standard furnishes a mechanism by which devices can ‘reach out’ to a Network Manager on their own (via a ‘trap’ message) to notify or alert the manager of a problem with the device. This typically requires each device on the network to be configured to issue SNMP traps to one or more network devices that are awaiting these traps.

The above message types are all encoded into messages referred to as Protocol Data Units (PDUs) that are interchanged between SNMP devices.

15.1.2 SNMP MIB Objects

The SNMP MIB is arranged in a tree-structured fashion, similar in many ways to a disk directory structure of files. The top level SNMP branch begins with the ISO ‘internet’ directory, which contains the following four main branches:

- **"mgmt" SNMP branch:** Contains the standard SNMP objects usually supported (at least in part) by all network devices.
- **"private" SNMP branch:** Contains those ‘extended’ SNMP objects defined by network equipment vendors.
- **"experimental" and "directory" SNMP branches:** Also defined within the ‘internet’ root directory, these branches are usually devoid of any meaningful data or objects.

The ‘tree’ structure described above is an integral part of the SNMP standard, though the most pertinent parts of the tree are the ‘leaf’ objects of the tree that provide actual management data regarding the device. Generally, SNMP leaf objects can be partitioned into two similar, but slightly different types that reflect the organization of the tree structure:

- **Discrete MIB Objects:** Contain one precise piece of management data. These objects are often distinguished from "Table" items (below) by adding a ".0" (dot-zero) extension to their names. The operator must merely know the name of the object and no other information.
- **Table MIB Objects:** Contain multiple sections of management data. These objects are distinguished from ‘Discrete’ items (above) by requiring a "." (dot) extension to their names that uniquely distinguishes the particular value being referenced. The "." (dot) extension is the "instance" number of an SNMP object. For "Discrete" objects, this instance number is zero. For "Table" objects, this instance number is the index into the SNMP table. SNMP tables are special types of SNMP objects which allow parallel arrays of information to be supported. Tables are distinguished from scalar objects, so that tables can grow without bounds. For example, SNMP defines the "ifDescr" object (as a standard SNMP object) that indicates the text description of each interface supported by a particular device. Since network devices can be configured with more than one interface, this object can only be represented as an array.

By convention, SNMP objects are always grouped in an "Entry" directory, within an object with a "Table" suffix. (The ifDescr object described above resides in the "ifEntry" directory contained in the "ifTable" directory).

15.1.3 SNMP Extensibility Feature

One of the principal components of an SNMP manager is a MIB Compiler which allows new MIB objects to be added to the management system. When a MIB is compiled into an SNMP manager, the manager is made "aware" of new objects that are supported by agents on the network. The concept is similar to adding a new schema to a database.

Typically, when a MIB is compiled into the system, the manager creates new folders or directories that correspond to the objects. These folders or directories can typically be

viewed with a MIB Browser, which is a traditional SNMP management tool incorporated into virtually all Network Management Systems.

The act of compiling the MIB allows the manager to know about the special objects supported by the agent and access these objects as part of the standard object set.

15.2 Carrier Grade Alarm System

The basic alarm system has been extended to a carrier-grade alarm system. A carrier-grade alarm system provides a reliable alarm reporting mechanism that takes into account EMS outages, network outages, and transport mechanism such as SNMP over UDP.

A carrier-grade alarm system is characterized by the following:

- The device has a mechanism that allows a manager to determine which alarms are currently active in the device. That is, the device maintains an active alarm table.
- The device has a mechanism to allow a manager to detect lost alarm raise and clear notifications [sequence number in trap, current sequence number MIB object].
- The device has a mechanism to allow a manager to recover lost alarm raise and clear notifications [maintains a log history].
- The device sends a cold start trap to indicate that it is starting. This allows the EMS to synchronize its view of the device's active alarms.

When the SNMP alarm traps are sent, the carrier-grade alarm system does not add or delete alarm traps as part of the feature. This system provides the mechanism for viewing of history and current active alarm information.

15.2.1 Active Alarm Table

The device maintains an active alarm table to allow a manager to determine which alarms are currently active in the device. Two views of the active alarm table are supported by the agent:

- acActiveAlarmTable in the enterprise acAlarm
- alarmActiveTable and alarmActiveVariableTable in the IETF standard ALARM-MIB (rooted in the AC tree)

The acActiveAlarmTable is a simple, one-row per alarm table that is easy to view with a MIB browser.

The ALARM-MIB is currently a draft standard and therefore has no OID assigned to it. In the current software release, the MIB is rooted in the experimental MIB subtree. In a future release, after the MIB has been ratified and an OID assigned, it is to move to the official OID.

15.2.2 Alarm History

The device maintains a history of alarms that have been raised and traps that have been cleared to allow a manager to recover any lost, raised or cleared traps. Two views of the alarm history table are supported by the agent:

- acAlarmHistoryTable in the enterprise acAlarm
- nlmLogTable and nlmLogVariableTable in the standard NOTIFICATION-LOG-MIB

As with the acActiveAlarmTable, the acAlarmHistoryTable is a simple, one-row-per-alarm table that is easy to view with a MIB browser.

15.3 Cold Start Trap

The gateway's technology supports a cold start trap to indicate that the device is starting. This allows the manager to synchronize its view of the device's active alarms. Two different traps are sent at start-up:

- The standard coldStart trap - iso(1).org(3).dod(6).internet(1). snmpV2(6). snmpModules(3). snmpMIB(1). snmpMIBObjects(1). snmpTraps(5). coldStart(1) - sent at system initialization.
- The enterprise acBoardEvBoardStarted which is generated at the end of system initialization. This is more of an ‘application-level’ cold start sent after the entire initializing process is complete and all the modules are ready.

15.4 Third-Party Performance Monitoring Measurements

Performance measurements are available for a third-party performance monitoring system (or EMS) through an SNMP interface. These measurements can be polled at scheduled intervals by an external poller or utility in a media server or other off-device system.

The device provides two types of performance measurements:

- **Gauges:** Gauges represent the current state of activities on the device. Gauges, unlike counters, can decrease in value, and like counters, can increase. The value of a gauge is the current value or a snapshot of the current activity on the device.
- **Counters:** Counters always increase in value and are cumulative. Counters, unlike gauges, never decrease in value unless the off-device system is reset. The counters are then zeroed.

Performance measurements are provided by several proprietary MIBs that are located under the acPerformance sub tree:

iso(1).org(3).dod(6).internet(1).private(4).enterprises(1).audioCodes(5003).acPerformance(10)

Two formats of performance monitoring MIBs are available:

- Old format (obsolete as of version 4.6):
  Each MIB is composed of a list of single MIB objects, each relates to a separate attribute within a gauge or a counter. All counters and gauges provide the current time value only.
  - **acPerfMediaGateway:** a generic-type of PM MIB that covers:
    - Control protocol
    - RTP stream
    - System packets statistics
  - **acPerfMediaServices:** media services devices specific performance MIB.
  - **acPerfH323SIPGateway:** holds statistics on Tel to IP and vice versa.
- New format:
  The following MIBs feature an identical structure. Each includes two major sub-trees.
  - Configuration sub tree – enables configuration of general attributes of the MIB and specific attributes of the monitored objects.
  - Data sub tree

The monitoring results are presented in tables. Each table includes one or two indices. When there are two indices, the first index is a sub-set in the table (e.g., trunk number) and the second (or a single where there is only one) index represents the interval number (present - 0, previous - 1 and the one before - 2).

The MIBs include:

- **acPMMedia:** for media (voice) related monitoring (e.g., RTP, DSP’s).
- **acPMControl:** for Control-Protocol related monitoring (e.g., connections, commands).

- **acPMPSTN:** for PSTN related monitoring such as channel use, trunk utilization. Note that the acPMTrunkUtilizationTable is not supported.
- **acPMSystem:** for general (system related) monitoring.
- The log trap, acPerformanceMonitoringThresholdCrossing (non-alarm) is sent every time the threshold of a Performance Monitored object is crossed. The severity field is "indeterminate" when the crossing is above the threshold and "cleared" when it falls bellow the threshold. The "source" varbind in the trap indicates the object for which the threshold is being crossed.

15.4.1 Total Counters

The counter’s attribute 'total' accumulates counter values since the board’s most recent restart. The user can reset the total’s value by setting the Reset-Total object.

Each MIB module has its own Reset Total object, as follows:

- **PM-Analog:** acPMAnalogConfigurationResetTotalCounters
- **PM-ATM:** acPMAtmConfigurationResetTotalCounters (not applicable to this release)
- **PM-Control:** acPMControlConfigurationResetTotalCounters
- **PM-Media:** acPMMediaConfigurationResetTotalCounters
- **PM-PSTN:** acPMPSTNConfigurationResetTotalCounters
- **PM-System:** acPMSystemConfigurationResetTotalCounters

15.5 TrunkPack-VoP Series Supported MIBs

The gateway contains an embedded SNMP Agent supporting the following MIBs:

- **Standard MIB (MIB-2):** The various SNMP values in the standard MIB are defined in RFC 1213. The standard MIB includes various objects to measure and monitor IP activity, TCP activity, UDP activity, IP routes, TCP connections, interfaces and general system indicators.
- **RTP MIB:** The RTP MIB is supported in conformance with the IETF RFC 2959. It contains objects relevant to the RTP streams generated and terminated by the device and to RTCP information related to these streams.
- **Trunk MIB:** The Trunk MIB contains objects relevant to E1/T1 Trunk interfaces.
- **NOTIFICATION-LOG-MIB:** This standard MIB (RFC 3014 - iso.org.dod.internet.mgmt.mib-2) is supported as part of our implementation of carrier grade alarms.
- **ALARM-MIB:** This is an IETF MIB (RFC 3877) also supported as part of our implementation of carrier grade alarms. This MIB is a new standard and is therefore under the audioCodes.acExperimental branch.
- **SNMP-TARGET-MIB:** According to RFC 2273. It allows for the configuration of trap destinations and trusted managers.
- **SNMP MIB:** This MIB (RFC 3418) allows support of the coldStart and authenticationFailure traps.
- **SNMP Framework MIB:** (RFC 3411).
- **SNMP Usm MIB:** this MIB (RFC 3414) implements the user-based Security Model.
- **SNMP Vacm MIB:** This MIB (RFC 3415) implements the view-based Access Control Model.
- **SNMP Community MIB:** This MIB (RFC 3584). implements community string management.

- **RTCP-XR:** This MIB (RFC) implements the following partial support:
  - The rtcpXrCallQualityTable is fully supported.
  - In the rtcpXrHistoryTable, support of the RCQ objects is provided only with no more than 3 intervals, 15 minutes long each.
  - Supports the rtcpXrVoipThresholdViolation trap.
- **ds1 MIB:** support for the following:
  - dsx1ConfigTable - partial supports following objects have SET and GET applied:
    - dsx1LineCoding
    - dsx1LoopbackConfig
    - dsx1LineStatusChangeTrapEnable
    - dsx1CircuitIdentifier

  All other objects in this table support GET only:
  - dsx1CurrentTable
  - dsx1IntervalTable
  - dsx1TotalTable
  - dsx1LineStatusChange trap
- **ipForward MIB:** (RFC 2096) fully supported

In addition to the standard MIBs, the complete product series contains proprietary MIBs:

- **AC-TYPES MIB:** lists the known types defined by the complete product series. This is referred to by the sysObjectID object in the MIB-II.

Note: The acBoard MIB is still supported but is being replaced by five newer proprietary MIBs. The only relevant section in this MIB is the trap sub-tree acTrap.

As noted above, five new MIBs cover the device's general parameters. Each contains a Configuration subtree for configuring related parameters. In some, there also are Status and Action subtrees.

The five MIBs are:

- **AC-CONTROL-MIB**
- **AC-MEDIA-MIB**
- **AC-PSTN-MIB**
- **AC-SYSTEM-MIB**
- **AC-SS7-MIB**
- **acGateway MIB:** This proprietary MIB contains objects related to configuration of the device when applied as a SIP or H.323 media gateway only. This MIB complements the other proprietary MIBs.

  The acGateway MIB has the following groups:
  - Common - for parameters common to both SIP and H.323
  - SIP - for SIP parameters only
  - H.323 - for H.323 parameters only
- **acAtm:** This proprietary MIB contains objects related to configuration and status of the device when applied as an ATM media gateway only. This MIB complements the other proprietary MIBs.
  The acAtm MIB has the following groups:
  - **acAtmConfiguration:** for configuring ATM related parameters
  - **acAtmStatus:** for the status of ATM connections

- **acAlarm:** This is AudioCodes' proprietary carrier-grade alarm MIB. It is a simpler implementation of the notificationLogMIB and the IETF suggested alarmMIB (both also supported in all AudioCodes' devices).
  The acAlarm MIB has the following groups:
  - **ActiveAlarm:** straightforward (single-indexed) table, listing all currently active alarms, together with their bindings (the alarm bindings are defined in acAlarm. acAlarmVarbinds and also in acBoard.acTrap. acBoardTrapDefinitions. oid_1_3_6_1_4_1_5003_9_10_1_21_2_0).
  - **acAlarmHistory:** straightforward (single-indexed) table, listing all recently raised alarms together with their bindings (the alarm bindings are defined in acAlarm. acAlarmVarbinds and also in acBoard.acTrap. acBoardTrapDefinitions. oid_1_3_6_1_4_1_5003_9_10_1_21_2_0).

The table size can be altered via notificationLogMIB.notificationLogMIBObjects.nlmConfig.nlmConfigGlobalEntryLimit or notificationLogMIB.notificationLogMIBObjects.nlmConfig.nlmConfigLogTable.nlm ConfigLogEntry.nlmConfigLogEntryLimit.

The table size can be any value between 50 and 1000 (default is 500).

Note 1: The following are special notes pertaining to MIBs:

- A detailed explanation of each parameter can be viewed in an SNMP browser in the "MIB Description" field.
- Not all groups in the MIB are functional. Refer to version release notes.
- Certain parameters are non-functional. Their MIB status is marked "obsolete".
- When a parameter is set to a new value via SNMP, the change may affect device functionality immediately or may require that the device be soft reset for the change to take effect. This depends on the parameter type.

Note 2: The current (updated) device configuration parameters are programmed into the device provided that the user does not load an *ini* file to the device after reset. Loading an *ini* file after reset overrides the updated parameters.

15.6 Traps

Full proprietary trap definitions and trap Varbinds are found in the acBoard MIB and acAlarm MIB. Table 15-1 lists the supported proprietary traps. For detailed information on these traps, refer to Appendix I on page 383.

Note: All traps are sent from the SNMP port (default 161). This is part of the NAT traversal solution.

Table 15-1: Proprietary Traps Description (continues on pages 314 to 315)

Trap	Description
acBoardFatalError	Sent whenever a fatal device error occurs.
acBoardConfigurationError	Sent when a device's settings are illegal. The trap contains a message describing the illegality of the setting. Note: Not applicable to IPM-260).
acBoardTemperatureAlarm	Sent when a board exceeds its temperature limits.
acBoardEvResettingBoard	Sent after the device is reset.
acBoardEvBoardStarted	Sent after the device is successfully restored and initialized following reset.
acgwAdminStateChange	Sent when Graceful Shutdown commences and ends.
acOperationalStateChange	Sent if the operational state of the node changes to disabled. Cleared when the operational state of the node changes to enabled.
acBoardCallResourcesAlarm	Sent when no free channels are available.
acBoardControllerFailureAlarm	Sent when the Gatekeeper/Proxy is not found or registration failed. Internal routing table can be used for routing.
acFeatureKeyError	Intended to relay Feature Key errors etc. (will be supported in the next applicable release).
acBoardOverloadAlarm	Sent when an overload in one or more of the system's components occurs.
acActiveAlarmTableOverflow	Sent to indicate that an active alarm could not be entered into the Active Alarm table because the table was full.
acKeepAlive	Part of the NAT traversal mechanism. If the STUN application detects a NAT, this trap is sent on regular time laps - 9/10 of the acSysSTUNBindingLifeTime object. The AdditionalInfo1 varbind has the MAC address of the gateway.
acNATTraversalAlarm	Sent when the NAT is placed in front of a gateway that is identified as a symmetric NAT. It is cleared when a non-symmetric NAT or no NAT replaces the symmetric one.

Table 15-1: Proprietary Traps Description (continues on pages 314 to 315)

Trap	Description
acEnhancedBITStatus	This trap is used to indicate the status of the Built In Test (BIT). The information in the trap contains board hardware elements being tested and their status. The information is presented in the additional info fields.
acBoardEthernetLinkAlarm	Sent when an Ethernet link(s) is down.
acPerformanceMonitoringThresholdCrossing	Sent every time the threshold of a Performance Monitored object is crossed. The severity field is 'indeterminate' when the crossing is above the threshold, and 'cleared' when it goes back under the threshold. The 'source' varbind in the trap indicates the object for which the threshold is being crossed.
acHTTPDownloadResult	Sent at the success or failure of HTTP download.
acDChannelStatus	Non-alarm trap sent at the establishment, re-establishment, or release of LAPD link with its peer connection. The trap is sent with one of the following textual descriptions: D-channel synchronized D-channel not-synchronized

In addition to the listed traps, the device also supports the following standard traps:

- dsx1LineStatusChange
- dsx3LineStatusChange
- coldStart
- authenticationFailure

15.7 SNMP Interface Details

This section describes details of the SNMP interface that is required when developing an Element Manager (EM) for any of the TrunkPack-VoP Series products, or to manage a device with a MIB browser.

The gateway offers the following SNMP security features:

- SNMPv2c community strings
- SNMPv3 User-based Security Model (USM) users
- SNMP encoded over IPSec (refer to Section 13.1 on page 281)
- Combinations of the above

Currently, both SNMP and *ini* file commands and downloads are not encrypted. For *ini* file encoding, refer to Section 6.1 on page 127.

15.7.1 SNMP Community Names

By default, the device uses a single, read-only community string of 'public' and a single read-write community string of 'private'.

The following community strings can be defined:

- Up to five read-only community strings
- Up to five read-write community strings
- A single trap community string

Each community string must be associated with one of the following predefined SNMP groups:

Table 15-2: SNMP Predefined Groups

Group	Get Access	Set Access	Can Send Traps
ReadGroup	Yes	No	Yes
ReadWriteGroup	Yes	Yes	Yes
TrapGroup	No	No	Yes

15.7.1.1 Configuration of Community Strings via the Web

For detailed information on configuration the community strings via the Embedded Web Server, refer to Section 5.6.9.2 on page 104.

15.7.1.2 Configuration of Community Strings via the *ini* File

The following *ini* file parameters are used to configure community strings:

- SNMPReadOnlyCommunityString_<x> = '#######'
- SNMPReadWriteCommunityString_<x> = '#######'

Where <x> is a number from 0 to 4. The '#' character represents any alphanumeric character. The maximum length of the string is 20 characters.

15.7.1.3 Configuration of Community Strings via SNMP

To configure read-only and read-write community strings, the EM must use the snmpCommunityMIB. To configure the trap community string, the EM must also use the snmpVacmMIB and the snmpTargetMIB.

➢ **To add a read-only community string (v2user), take these 2 steps:**

1. Add a new row to the snmpCommunityTable with CommunityName v2user.
2. Add a row to the vacmSecurityToGroupTable for SecurityName v2user, GroupName ReadGroup and SecurityModel snmpv2c.

➢ **To delete the read-only community string (v2user), take these 3 steps:**

1. If v2user is being used as the trap community string, follow the procedure for changing the trap community string. (See below.)
2. Delete the snmpCommunityTable row with CommunityName v2user.
3. Delete the vacmSecurityToGroupTable row for SecurityName v2user, GroupName ReadGroup and SecurityModel snmpv2c.

➢ To add a read-write community string (v2admin), take these 2 steps:

1. Add a new row to the snmpCommunityTable with CommunityName v2admin.
2. Add a row to the vacmSecurityToGroupTable for SecurityName v2admin, GroupName ReadWriteGroup and SecurityModel snmpv2c.

➢ To delete the read-write community string (v2admin), take these 2 steps:

1. If v2admin is being used as the trap community string, follow the procedure for changing the trap community string. (Refer to the procedure below.)
2. Delete the snmpCommunityTable row with a CommunityName of v2admin and GroupName of ReadWriteGroup.

➢ To change the only read-write community string from v2admin to v2mgr, take these 4 steps:

1. Follow the procedure above to add a read-write community string to a row for v2mgr.
2. Set up the EM so that subsequent 'set' requests use the new community string, v2mgr.
3. If v2admin is being used as the trap community string, follow the procedure to change the trap community string (see below).
4. Follow the procedure above to delete a read-write community name in the row for v2admin.

➢ To change the trap community string, take these 3 steps:

1. Add a row to the vacmSecurityToGroupTable with these values:
   - SecurityModel = 2
   - SecurityName = the new trap community string
   - GroupName = TrapGroup, ReadGroup or ReadWriteGroup
   - The SecurityModel and SecurityName objects are row indices
2. Modify the SecurityName field in the sole row of the snmpTargetParamsTable.
3. Remove the row from the vacmSecurityToGroupTable with SecurityName = the old trap community string.

Notes:

- The procedure above assumes that a row already exists in the snmpCommunityTable for the new trap community string. The trap community string can be part of the TrapGroup, ReadGroup, or ReadWriteGroup. If the trap community string is used solely for sending traps (recommended), then it should be made part of the TrapGroup.
- You must add GroupName and RowStatus on the same set.

15.7.2 SNMP v3 USM Users

You can define up to 10 User-based Security Model (USM) users (USM users are referred to as “v3 users”). Each v3 user can be associated with an authentication type (none, MD5, or SHA-1) and a privacy type (none, DES, 3DES, or AES).

Table 15-3: SNMP v3 Security Levels

Security Level	Authentication	Privacy
noAuthNoPriv(1)	None	None
authNoPriv(2)	MD5 or SHA-1	None
authPriv(3)	MD5 or SHA-1	DES, 3DES, AES128, AES192, or AES256

Each SNMP v3 user must be associated with one of the predefined groups listed in the following table:

Table 15-4: SNMP v3 Predefined Groups

Group	Get Access	Set Access	Send Traps	Security Level
ReadGroup1	Yes	No	Yes	noAuthNoPriv(1)
ReadWriteGroup1	Yes	Yes	Yes	noAuthNoPriv(1)
TrapGroup1	No	No	Yes	noAuthNoPriv(1)
ReadGroup2	Yes	No	Yes	authNoPriv(2)
ReadWriteGroup2	Yes	Yes	Yes	authNoPriv(2)
TrapGroup2	No	No	Yes	authNoPriv(2)
ReadGroup3	Yes	No	Yes	authPriv(3)
ReadWriteGroup3	Yes	Yes	Yes	authPriv(3)
TrapGroup3	No	No	Yes	authPriv(3)

15.7.2.1 Configuring SNMP v3 users via the *ini* File

Use the SNMPUsers *ini* table to add, modify, and delete SNMPv3 users. For a description of the SNMPUsers table *ini* file parameters, refer to Section 6.11 on page 148.

Note: The SNMPUsers *ini* table is a hidden parameter. Therefore, when you perform a “Get ini File” operation using the Web interface, the table will not be included in the generated file.

You can enter keys in the form of a text password or in the form of a localized key in hex format. If using a text password, then it should be at least eight characters in length. Below is an example of a localized key format:

```
26:60:d8:7d:0d:4a:d6:8c:02:73:dd:22:96:a2:69:df
```

The following example configuration creates three SNMPv3 USM users:

```
[ SNMPUsers ]
FORMAT SNMPUsers_Index = SNMPUsers_Username, SNMPUsers_AuthProtocol,
SNMPUsers_PrivProtocol, SNMPUsers_AuthKey, SNMPUsers_PrivKey, SNMPUsers_Group;
SNMPUsers 0 = v3user, 0, 0, -, -, 0;
SNMPUsers 1 = v3admin1, 1, 0, myauthkey, -, 1;
SNMPUsers 2 = v3admin2, 2, 1, myauthkey, myprivkey, 1;
[ \SNMPUsers ]
```

The example above creates the following three v3 users:

- The user "v3user" is defined for a security level of noAuthNoPriv(1) and is associated with ReadGroup1.
- The user "v3admin1" is defined for a security level of authNoPriv(2) with authentication protocol MD5. The authentication text password is “myauthkey” and the user will be associated with ReadWriteGroup2.
- The user "v3admin2" is defined for a security level of authPriv(3) with authentication protocol SHA-1 and privacy protocol DES. The authentication text password is “myauthkey”, the privacy text password is “myprivkey”, and the user will be associated with ReadWriteGroup3.

15.7.2.2 Configuring SNMP v3 Users via SNMP

To configure SNMP v3 users, the EM must use the standard snmpUsmMIB and the snmpVacmMIB.

➢ To add a read-only, noAuthNoPriv SNMPv3 user (v3user), take these 3 steps:

1. Clone the row with the same security level. After the clone step, the status of the row is notReady(3).
2. Activate the row (i.e., set the row status to active(1)).
3. Add a row to the vacmSecurityToGroupTable for SecurityName v3user, GroupName ReadGroup1, and SecurityModel usm(3).

Note: A row with the same security level (noAuthNoPriv) must already exist in the usmUserTable. (See the usmUserTable for details).

➢ To delete the read-only, noAuthNoPriv SNMPv3 user (v3user), take these 3 steps:

1. If v3 user is associated with a trap destination, follow the procedure for associating a different user to that trap destination. (See below.)
2. Delete the vacmSecurityToGroupTable row for SecurityName v3user, GroupName ReadGroup1, and SecurityModel usm.
3. Delete the row in the usmUserTable for v3user.

➢ To add a read-write, authPriv SNMPv3 user (v3user), take these 4 steps:

1. Clone the row with the same security level.
2. Change the authentication key and privacy key.
3. Activate the row. That is, set the row status to active(1).
4. Add a row to the vacmSecurityToGroupTable for SecurityName v3admin1, GroupName ReadWriteGroup3, and SecurityModel usm(3).

Note: A row with the same security level (authPriv) must already exist in the usmUserTable (see the usmUserTable for details).

➢ To delete the read-write, authPriv SNMPv3 user (v3admin1), take these 3 steps:

1. If v3admin1 is associated with a trap destination, follow the procedure for associating a different user to that trap destination. (See below.)
2. Delete the vacmSecurityToGroupTable row for SecurityName v3admin1, GroupName ReadWriteGroup1, and SecurityModel usm.
3. Delete the row in the usmUserTable for v3admin1.

15.7.3 Trusted Managers

By default, the agent accepts ‘get’ and ‘set’ requests from any IP address, as long as the correct community string is used in the request. Security can be enhanced via the use of Trusted Managers. A Trusted Manager is an IP address from which the SNMP Agent accepts and processes ‘get’ and ‘set’ requests. An EM can be used to configure up to five Trusted Managers.

Note: If Trusted Managers are defined, all community strings work from all Trusted Managers. That is, there is no way to associate a community string with particular trusted managers.

The concept of trusted managers is considered to be a weak form of security and is therefore, not a required part of SNMPv3 security, which uses authentication and privacy. However, the board’s SNMP agent applies the trusted manager concept as follows:

- There is no way to configure trusted managers for only a SNMPv3 user. An SNMPv2c community string must be defined.
- If specific IPs are configured as trusted managers (via the community table), then only SNMPv3 users on those trusted managers are given access to the agent’s MIB objects.

15.7.3.1 Configuration of Trusted Managers via *ini* File

To set the Trusted Mangers table from start-up, write the following in the *ini* file:

SNMPTRUSTEDMGR_X = D.D.D.D

where X is any integer between 0 and 4 (0 sets the first table entry, 1 sets the second, and so on), and D is an integer between 0 and 255.

15.7.3.2 Configuration of Trusted Managers via SNMP

To configure Trusted Managers, the EM must use the SNMP-COMMUNITY-MIB, the snmpTargetMIB and the snmpTargetMIB.

The procedure below assumes that there is at least one configured read-write community, are currently no Trusted Managers, and the TransportTag for columns for all snmpCommunityTable rows are currently empty.

➢ To add the first Trusted Manager, take these 3 steps:

1. Add a row to the snmpTargetAddrTable with these values:
   - Name=mgr0
   - TagList=MGR
   - Params=v2cparams
2. Add a row to the snmpTargetAddrExtTable table with these values:
   - Name=mgr0
   - snmpTargetAddrTMask=255.255.255.255:0.

   The agent does not allow creation of a row in this table unless a corresponding row exists in the snmpTargetAddrTable.
3. Set the value of the TransportTag field on each non-TrapGroup row in the snmpCommunityTable to MGR.

The following procedure assumes that there is at least one configured read-write community, are currently one or more Trusted Managers, and the TransportTag for columns for all rows in the snmpCommunityTable are currently set to MGR. This procedure must be done from one of the existing Trusted Managers

➢ To add a subsequent Trusted Manager, take these 2 steps:

1. Add a row to the snmpTargetAddrTable with these values:
   - Name=mgrN
   - TagList=MGR
   - Params=v2cparams

   Where N is an unused number between 0 and 4.
2. Add a row to the snmpTargetAddrExtTable table with these values:
   - Name=mgrN
   - snmpTargetAddrTMask=255.255.255.255:0

An alternative to the above procedure is to set the snmpTargetAddrTMask column while you are creating other rows in the table.

The procedure below assumes that there is at least one configured read-write community, are currently two or more Trusted Managers, and the taglist for columns for all rows in the snmpCommunityTable are currently set to MGR. This procedure must be done from one of the existing trusted managers, but not the one that is being deleted.

➢ To delete a Trusted Manager (not the final one), take this step:

- Remove the appropriate row from the snmpTargetAddrTable.

The change takes affect immediately. The deleted trusted manager cannot access the board. The agent automatically removes the row in the snmpTargetAddrExtTable.

The procedure below assumes that there is at least one configured read-write community, currently only one Trusted Manager, and the taglist for columns for all rows in the snmpCommunityTable are currently set to MGR. This procedure must be done from thefinal Trusted Manager

➢ To delete the final Trusted Manager, take these 2 steps:

1. Set the value of the TransportTag field on each row in the snmpCommunityTable to the empty string.
2. Remove the appropriate row from the snmpTargetAddrTable.

The change takes affect immediately. All managers can now access the board. The agent automatically removes the row in the snmpTargetAddrExtTable.

15.7.4 SNMP Ports

The SNMP Request Port is 161 and the Trap Port is 162. These ports can be changed by setting parameters in the device *ini* file. The parameter name is:

SNMPPort = <port_number>
Valid UDP port number; default = 161

This parameter specifies the port number for SNMP requests and responses. Usually, it should not be specified. Use the default.

15.7.5 Multiple SNMP Trap Destinations

An agent can send traps to up to five managers. For each manager, set the manager's IP address, receiving port number, and enable sending traps to that manager.

The user also has the option of associating a trap destination with a specific SNMPv3 USM user. Traps are then sent to that trap destination using the SNMPv3 format and the authentication and privacy protocol configured for that user..

To configure the trap managers table use:

- The Embedded Web Server (refer to Section 5.6.9.1 on page 103).
- The *ini* file (refer to Section 15.7.5.2 below).
- SNMP (refer to Section 15.7.5.3 on page 324).

15.7.5.1 Configuring Trap Manager via Host Name

One of the five available SNMP managers can be defined using a FQDN. In the current version, this option can only be configured via the *ini* file (SNMPTrapManagerHostName).

The gateway tries to resolve the host name at start up. Once the name is resolved (IP is found), the resolved IP address replaces the last entry in the trap manager table (defined by the parameter SNMPManagerTableIP_x) and the last trap manager entry of snmpTargetAddrTable in the snmpTargetMIB. The port is 162 (unless specified otherwise), the row is marked as 'used' and the sending is 'enabled'.

When using 'host name' resolution, any changes made by the user to this row in either MIBs are overwritten by the gateway when a resolving is redone (once an hour).

Note that several traps may be lost until the resolving is complete.

15.7.5.2 Configuring Trap Managers via the ini File

In the IPmedia 2000 *ini* file, the parameters below can be set to enable or disable the sending of SNMP traps. Multiple trap destinations can be supported on the device by setting multiple trap destinations in the *ini* file.

- **SNMPManagerTrapSendingEnable_<x>:** indicates whether or not traps are to be sent to the specified SNMP trap manager. A value of '1' means that it is enabled, while a value of '0' means disabled. <x> represents a number 0, 1, 2 which is the array element index. Currently, up to five SNMP trap managers can be supported.
- **SNMPManagerTrapUser_<x>**: indicates to send an SNMPv2 trap using the trap user community string configured with the SNMPTrapCommunityString parameter. The user may instead specify an SNMPv3 user name.

Figure 15-1 presents an example of entries in a device's *ini* file regarding SNMP. The device can be configured to send to multiple trap destinations. The lines in the file below are commented with the ';' at the beginning of the line. All of the lines below are commented since the first line character is a semi-colon.

Figure 15-1: Example of Entries in a Device *ini* file Regarding SNMP

```
; SNMP trap destinations
; The board maintains a table of trap destinations containing 5 ;rows. The rows
are numbered 0..4. Each block of 4 items below ;apply to a row in the table.

; To configure one of the rows, uncomment all 4 lines in that ;block. Supply an
IP address and if necessary, change the port ;number.
; To delete a trap destination, set ISUSED to 0.
; -change these entries as needed
;SNMPManagerTableIP_0=
;SNMPManagerTrapPort_0=162
;SNMPManagerIsUsed_0=1
;SNMPManagerTrapSendingEnable_0=1
;
;SNMPManagerTableIP_1=
;SNMPManagerTrapPort_1=162
;SNMPManagerIsUsed_1=1
;SNMPManagerTrapSendingEnable_1=1
;
;SNMPManagerTableIP_2=
;SNMPManagerTrapPort_2=162
;SNMPManagerIsUsed_2=1
;SNMPManagerTrapSendingEnable_2=1
;
;SNMPManagerTableIP_3=
;SNMPManagerTrapPort_3=162
;SNMPManagerIsUsed_3=1
;SNMPManagerTrapSendingEnable_3=1
;
;SNMPManagerTableIP_4=
;SNMPManagerTrapPort_4=162
;SNMPManagerIsUsed_4=1
;SNMPManagerTrapSendingEnable_4=1
```

To configure the trap manger host name, use the parameter SNMPTrapManagerHostName. For example:

```
;SNMPTrapManagerHostName = 'myMananger.corp.MyCompany.com'
```

Note: The same information configurable in the *ini* file can also be configured via the acBoardMIB.

15.7.5.3 Configuring Trap Managers via SNMP

The standard snmpTargetMIB interface is available for configuring trap managers.

> **Note:** The acBoard MIB is planned to become obsolete. The only relevant section in this MIB is the trap sub tree acTrap.

➢ To add an SNMPv2 trap destination, take this step:

- Add a row to the snmpTargetAddrTable with these values:
  - Name=trapN (where N is an unused number between 0 and 4)
  - TagList=AC_TRAP
  - Params=v2cparams

  All changes to the trap destination configuration take effect immediately.

➢ To add an SNMPv3 trap destination, take these 2 steps:

1. Add a row to the snmpTargetAddrTable with these values:
   - Name=trapN, where N is an unused number between 0 and 4
   - TagList=AC_TRAP
   - Params=usm<user>, where <user> is the name of the SNMPv3 with which this user is associated
2. If a row does not already exist for this combination of user and SecurityLevel, add a row to the snmpTargetParamsTable with these values:
   - Name=usm<user>
   - MPModel=3(SNMPv3)
   - SecurityModel=3 (usm)
   - SecurityName=<user>
   - SecurityLevel=M, where M is either 1(noAuthNoPriv), 2(authNoPriv), or 3(authPriv)

All changes to the trap destination configuration take effect immediately.

➢ To delete a trap destination, take this step:

- Remove the appropriate row from the snmpTargetAddrTable.

If this is the last trap destination associated with this user and security level, you can also delete the appropriate row from the snmpTargetParamsTable.

You can change the IP address and/or port number of an existing trap destination. The same effect can be achieved by removing a row and adding a new row.

➢ To modify a trap destination, take this step:

- Modify the IP address and/or port number for the appropriate row in the snmpTargetAddrTable.

➢ To disable a trap destination, take this step:

- Change TagList on the appropriate row in the snmpTargetAddrTable to the empty string.

➢ To enable a trap destination, take this step:

- Change TagList on the appropriate row in the snmpTargetAddrTable to ‘AC_TRAP’.

15.8 SNMP Manager Backward Compatibility

With support for the Multi Manager Trapping feature, the older acSNMPManagerIP MIB object, which is synchronized with the first index in the snmpManagers MIB table is also supported. This is translated in two features:

- SET/GET to either of the two MIB objects is identical.
  In other words, as far as the SET/GET are concerned, OID 1.3.6.1.4.1.5003.9.10.1.1.2.7 is identical to OID 1.3.6.1.4.1.5003.9.10.1.1.2.21.1.1.3.
- When setting ANY IP to the acSNMPManagerIP (this is the older parameter, not the table parameter), two more parameters are SET to ENABLE. snmpManagerIsUsed.0 and snmpManagerTrapSendingEnable.0 are both set to 1.

15.9 Dual Module Interface

Dual module boards comprise of two modules (the first module is on the right side of the board when looking at it from the front). As the two modules reside in a single board it is important to differentiate between them. Differentiation is based on the modules’ serial numbers.

MIB object acSysIdSerialNumber always returns the serial number of the module on which the GET operation is performed.

MIB object acSysIdFirstSerialNumber always returns the serial number of the first module.

If the module on which the GET operation is performed is the second module, then the values of these two objects are different. If, on the other hand, the module is the first module, the values of these two objects are the same.

15.10 SNMP NAT Traversal

A NAT placed between the gateway and the element manager calls for traversal solutions:

- **Trap source port:** all traps are sent out from the SNMP port (default 161). A manager receiving these traps can use the binding information (in the UDP layer) to traverse the NAT back to the device.
  The trap destination address (port and IP) are as configured in the snmpTargetMIB.
- **acKeepAliveTrap:** this trap is designed to be a constant life signal from the device to the manager allowing the manager NAT traversal at all times. The acBoardTrapGlobalsAdditionalInfo1 varbind has the device's serial number.
  The destination port (i.e., the manager port for this trap) can be set to be different than the port to which all other traps are sent. To do this, use the acSysSNMPKeepAliveTrapPort object in the acSystem MIB or the *ini* file parameter KeepAliveTrapPort.
  The Trap is instigated in three ways:
  - Via an *ini* file parameter (SendKeepAliveTrap = 1). This ensures that the trap is continuously sent. The frequency is set via the 9/10 of the acSysSTUNBindingLifeTime object.
  - After the STUN client has discovered a NAT (any NAT).
  - If the STUN client cannot contact a STUN server.

> **Note:** The two latter options require the STUN client be enabled (EnableSTUN). Also, once the acKeepAlive trap is instigated it does not stop.

- The manager can view the NAT type in the MIB:
  audioCodes(5003).acProducts(9).acBoardMibs(10).acSystem(10).acSystemStatus(2).acSysNetwork(6).acSysNAT(2).acSysNATType(1)
- The manger also has access to the STUN client configuration:
  audioCodes(5003).acProducts(9).acBoardMibs(10).acSystem(10).acSystemConfiguration(1).acSysNetworkConfig(3).acSysNATTraversal(6).acSysSTUN(21)
- **acNATTraversalAlarm:** When the NAT is placed in front a device that is identified as a symmetric NAT, this alarm is raised. It is cleared when a non-symmetric NAT or no NAT replace the symmetric one.

15.11 SNMP Administrative State Control

15.11.1 Node Maintenance

Node maintenance for the IPmedia 2000 is provided by an SNMP interface. The acBoardMIB provides two parameters for graceful and forced shutdowns of the IPmedia 2000:

- acgwAdminState
- acgwAdminStateLockControl

The acgwAdminState is used either to request (set) a shutdown (0), undo shutdown (2), or to view (get) the gateway condition (0 = locked; 1 = shutting down; 2 = unlocked).

The acgwAdminStateLockControl is used to set a time limit (in seconds) for the shutdown where 0 means shutdown immediately (forced), -1 means no time limit (graceful), and x where x>0 indicates a time limit in seconds (timed limit is considered a graceful shutdown).

Note: The acgwAdminStateLockControl must be set first followed by the acgwAdminState.

15.11.2 Graceful Shutdown

acgwAdminState is a read-write MIB object. When a get request is sent for this object, the agent returns the board's current administrative state.

The possible values received on a **get** request include the following:

- locked(0): the board is locked
- shuttingDown(1): the board is currently performing a graceful lock
- unlocked(2): the board is unlocked

On a **set** request, the manager supplies one of the following desired administrative states:

- locked(0)
- unlocked(2)

When the board changes to either shuttingDown or locked state, an adminStateChange alarm is raised. When the board changes to an unlocked state, the adminStateChange alarm is cleared.

Before setting acgwAdminState to perform a lock, acgwAdminStateLockControl should be set first to control the type of lock that is performed. The possible values for the acgwAdminStateLockControl include the following:

- 1 = Perform a graceful lock. Calls are allowed to complete. No new calls are allowed to be originated on this device.
- 0 = Perform a forced lock. Calls are immediately terminated.
- Any number greater than 0 = Time in seconds before the graceful lock turns into a forced lock.

15.12 AudioCodes' Element Management System

Using AudioCodes' Element Management System (EMS) is recommended to Customers requiring large deployments (multiple media gateways in globally distributed enterprise offices, for example), that need to be managed by central personnel.

The EMS is not included in the device's supplied package. Contact AudioCodes for detailed information on AudioCodes' EMS and on AudioCodes' EVN - Enterprise VoIP Network – solution for large VoIP deployments.

Note: The EMS doesn't apply to the TP-260.

16 Configuration Files

This section describes the configuration (*dat*) files that are load (in addition to the *ini* file) to the gateway. The configuration files are:

- Call Progress Tones file (refer to Section 16.1 below).
- Prerecorded Tones file (refer to Section 16.2on page 332).
- Voice Prompts file (refer to Section 16.3 on page 332).
- CAS protocol configuration files (refer to Section 16.4 on page 333).
- User Information file (refer to Section 16.5 on page 333).

To load any of the configuration files to the gateway use the Embedded Web Server (refer to Section 5.8.2 on page 119) or alternatively specify the name of the relevant configuration file in the gateway's *ini* file and load it (the *ini* file) to the gateway (refer to Section D.6 on page 354).

16.1 Configuring the Call Progress Tones

The Call Progress Tones, configuration file used by the gateway is a binary file (with the extension *dat*) which contains the definitions of the Call Progress Tones (levels and frequencies) that are detected / generated by the gateway.

Users can either use, one of the supplied gateway configuration (*dat*) files, or construct their own file. To construct their own configuration file, users are recommended, to modify the supplied *usa_tone.ini* file (in any standard text editor) to suit their specific requirements, and to convert it (the modified *ini* file) into binary format using the TrunkPack Downloadable Conversion Utility. For the description of the procedure on how to convert CPT *ini* file to a binary *dat* file, refer to Section G.1.1 on page 370.

To load the Call Progress Tones *(dat*) file to the gateway, use the Embedded Web Server (refer to Section 5.8.2 on page 119) or the *ini* file (refer to Section 6.18 on page 201).

Note: Only the *dat* file can be loaded to the gateway.

16.1.1 Format of the Call Progress Tones Section in the *ini* File

Users can create up to 32 different Call Progress Tones, each with frequency and format attributes.

The frequency attribute can be single or dual-frequency (in the range of 300 Hz to 1980 Hz), or an Amplitude Modulated (AM). In total, up to 64 different frequencies are supported. Only eight AM tones, in the range of 1 to 128 kHz, can be configured (the detection range is limited to 1 to 50 kHz). Note that when a tone is composed of a single frequency, the second frequency field must be set to zero.

The format attribute can be one of the following:

- **Continues:** (e.g., dial tone) a steady non-interrupted sound. Only the 'First Signal On time' should be specified. All other on and off periods must be set to zero. In this case, the parameter specifies the detection period. For example, if it equals 300, the tone is detected after 3 seconds (300 x 10 msec). The minimum detection time is 100 msec.
- **Cadence:** A repeating sequence of on and off sounds. Up to four different sets of on / off periods can be specified.

- **Burst:** A single sound followed by silence. Only the 'First Signal On time' and 'First Signal Off time' should be specified. All other on and off periods must be set to zero. The burst tone is detected after the off time is completed.

Users can specify several tones of the same type. These additional tones are used only for tone detection. Generation of a specific tone conforms to the first definition of the specific tone. For example, users can define an additional dial tone by appending the second dial tone's definition lines to the first tone definition in the *ini* file. The gateway reports dial tone detection if either of the two tones is detected.

The Call Progress Tones section of the *ini* file comprises the following segments:

- **[NUMBER OF CALL PROGRESS TONES]:** Contains the following key:
  - 'Number of Call Progress Tones' defining the number of Call Progress Tones that are defined in the file.
- **[CALL PROGRESS TONE #X]:** containing the Xth tone definition (starting from 1 and not exceeding the number of Call Progress Tones defined in the first section) using the following keys:
  - **Tone Type:** Call Progress Tone type

Figure 16-1: Call Progress Tone Types

```
1.   Dial Tone
2.   Ringback Tone
3.   Busy Tone
7.   Reorder Tone
17.Call Waiting Ringback Tone
23. Hold Tone
```

  - **Tone Modulation Type:** Either Amplitude Modulated (1) or regular (0).
  - **Tone Form:** The tone's format, can be one of the following:
    - Continuous
    - Cadence
    - Burst
  - **Low Freq [Hz]:** Frequency in hertz of the lower tone component in case of dual frequency tone, or the frequency of the tone in case of single tone (not relevant to AM tones).
  - **High Freq [Hz]:** Frequency in hertz of the higher tone component in case of dual frequency tone, or zero (0) in case of single tone (not relevant to AM tones).
  - **Low Freq Level [-dBm]:** Generation level 0 dBm to –31 dBm in [dBm] (not relevant to AM tones).
  - **High Freq Level:** Generation level. 0 to –31 dBm. The value should be set to '32' in the case of a single tone (not relevant to AM tones).
  - **First Signal On Time [10 msec]:** 'Signal On' period (in 10 msec units) for the first cadence on-off cycle. For be continuous tones, this parameter defines the detection period. For burst tones, it defines the tone's duration.
  - **First Signal Off Time [10 msec]:** 'Signal Off' period (in 10 msec units) for the first cadence on-off cycle (for cadence tones). For burst tones, this parameter defines the off time required after the burst tone ends and the tone detection is reported. For continuous tones, this parameter is ignored.
  - **Second Signal On Time [10 msec]:** 'Signal On' period (in 10 msec units) for the second cadence on-off cycle. Can be omitted if there isn't a second cadence.
  - **Second Signal Off Time [10 msec]:** 'Signal Off' period (in 10 msec units) for the second cadence on-off cycle. Can be omitted if there isn't a second cadence.
  - **Third Signal On Time [10 msec]:** 'Signal On' period (in 10 msec units) for the third cadence ON-OFF cycle. Can be omitted if there isn't a third cadence.

- **Third Signal Off Time [10 msec]:** ‘Signal Off’ period (in 10 msec units) for the third cadence ON-OFF cycle. Can be omitted if there isn’t a third cadence.
- **Forth Signal On Time [10 msec]:** ‘Signal On’ period (in 10 msec units) for the fourth cadence ON-OFF cycle. Can be omitted if there isn’t a fourth cadence.
- **Forth Signal Off Time [10 msec]:** ‘Signal Off’ period (in 10 msec units) for the fourth cadence ON-OFF cycle. Can be omitted if there isn’t a fourth cadence.
- **Carrier Freq [Hz]:** the frequency of the carrier signal for AM tones.
- **Modulation Freq [Hz]:** the frequency of the modulated signal for AM tones (valid range from 1 Hz to 128 Hz).
- **Signal Level [-dBm]:** the level of the tone for AM tones.
- **AM Factor [steps of 0.02]:** the amplitude modulation factor (valid range from 1 to 50. Recommended values from 10 to 25).

Notes:

- When the same frequency is used for a continuous tone and a cadence tone, the ‘Signal On Time’ parameter of the continuous tone must have a value that is greater than the ‘Signal On Time’ parameter of the cadence tone. Otherwise the continuous tone is detected instead of the cadence tone.
- The tones frequency should differ by at least 40 Hz from one tone to other defined tones.

For example: to configure the dial tone to 440 Hz only, define the following text:

Figure 16-2: Defining a Dial Tone Example

```
#Dial tone
[CALL PROGRESS TONE #1]
Tone Type=1
Tone Form =1 (continuous)
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=10 (-10 dBm)
High Freq Level [-dBm]=32 (use 32 only if a single tone is required)
First Signal On Time [10msec]=300; the dial tone is detected after 3 sec
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=0
Second Signal Off Time [10msec]=0
```

Figure 16-3: Example of Ringing Burst

```
#Three ringing bursts followed by repeated ringing of 1 sec on and 3 sec off.
[NUMBER OF DISTINCTIVE RINGING PATTERNS]
Number of Ringing Patterns=1
[Ringing Pattern #0]
Ring Type=0
Freq [Hz]=25
First Burst Ring On Time [10msec]=30
First Burst Ring Off Time [10msec]=30
Second Burst Ring On Time [10msec]=30
Second Burst Ring Off Time [10msec]=30
Third Burst Ring On Time [10msec]=30
Third Burst Ring Off Time [10msec]=30
Fourth Ring On Time [10msec]=100
Fourth Ring Off Time [10msec]=300
```

16.2 Prerecorded Tones (PRT) File

The Call Progress Tones mechanism has several limitations, such as a limited number of predefined tones and a limited number of frequency integrations in one tone. To work around these limitations and provide tone generation capability that is more flexible, the PRT file can be used. If a specific prerecorded tone exists in the PRT file, it takes precedence over the same tone that exists in the CPT file and is played instead of it.

Note that the prerecorded tones are used only for generation of tones. Detection of tones is performed according to the CPT file.

16.2.1 PRT File Format

The PRT *dat* file contains a set of prerecorded tones to be played by the gateway during operation. Up to 40 tones (totaling approximately 10 minutes) can be stored in a single file in flash memory. The prerecorded tones (raw data PCM or L8 files) are prepared offline using standard recording utilities (such as CoolEdit™) and combined into a single file using the TrunkPack Downloadable Conversion utility (refer to Section G.1.5 on page 375).

The raw data files must be recorded with the following characteristics:

- Coders: G.711 A-law, G.711 µ-law or Linear PCM
- Rate: 8 kHz
- Resolution: 8-bit
- Channels: mono

The generated PRT file can then be loaded to the gateway using the BootP/TFTP utility (refer to Section 6.18 on page201) or via the Embedded Web Server (Section 5.8.2 on page 119).

The prerecorded tones are played repeatedly. This enables you to record only part of the tone and play it for the full duration. For example, if a tone has a cadence of 2 seconds on and 4 seconds off, the recorded file should contain only these 6 seconds. The PRT module repeatedly plays this cadence for the configured duration. Similarly, a continuous tone can be played by repeating only part of it.

16.3 Voice Prompts File

The voice announcement file contains a set of Voice Prompts to be played by the gateway during operation. The voice announcements are prepared offline using standard recording utilities and combined into a single file using the TrunkPack Downloadable Conversion Utility.

The generated announcement file can then be loaded to the gateway using the BootP/TFTP utility (refer to Section 6.18 on page 201) or via the Embedded Web Server (Section 5.8.2 on page 119).

If the size of the combined Voice Prompts file is less than 1 MB, it can permanently be stored in flash memory. Larger files, up to 10 MB, are stored in RAM, and should be loaded again (using BootP/TFTP utility) after the gateway is reset.

The Voice Prompts integrated file is a collection of raw voice recordings and / or *wav* files. These recordings can be prepared using standard utilities, such as CoolEdit, Goldwave™ and others. The raw voice recordings must be sampled at 8000 kHz / mono / 8 bit. The *wav* files must be recorded with G.711µ-Law/A-Law/Linear.

When the list of recorded files is converted to a single *voiceprompts.dat* file, every Voice Prompt is tagged with an ID number, starting with '1'. This ID is used later by the gateway to start playing the correct announcement. Up to 1000 Voice Prompts can be used.

AudioCodes provides a professionally recorded English (U.S.) Voice Prompts file.

➢ To generate and load the Voice Prompts file, take these 3 steps:

1. Prepare one or more voice files using standard utilities.
2. Use the TrunkPack Downloadable Conversion Utility to generate the *voiceprompts.dat* file from the pre-recorded voice messages (refer to Section G.1.2 on page 371).
3. Load the *voiceprompts.dat* file to the gateway either by using a TFTP procedure (refer to Section 6.18 on page 201), or via the Embedded Web Server (Section 5.8.2 on page 119).

16.4 CAS Protocol Configuration Files

The CAS Protocol Configuration Files contain the CAS Protocol definitions to be used for CAS-terminated trunks, users can either use the files supplied or construct their own files.

It is possible to load up to eight files and to use different files for different trunks.

Note that all CAS files loaded together must belong to the same Trunk Type (either E1 or T1).

16.5 User Information File

The User Information file maps PBX extensions to global IP numbers. In this context, a global IP number serves as a routing identifier for calls in the 'IP World'. The PBX extension uses this mapping to emulate the behavior of an IP phone. Note that the mapping mechanism is disabled by default and must be activated using the parameter 'EnableUserInfoUsage' (described in Section 6.12 on page 150).

Each line in the file represents a mapping rule of a single PBX extension (up to 1000 rules can be configured). Each line includes five items separated with commas. The items are described in Table 16-1 below. An example of a User Information file is shown in Figure 16-4 below.

Each PBX extension registers (and authenticated) separately (a REGISTER message is sent for each entry, only if AuthenticationMode is set to 'Per Endpoint') using the IP number in the From / To headers. The REGISTER messages are sent gradually (i.e., initially, the gateway sends requests according to the maximum number of allowed SIP dialogs (configured by the parameter NumberOfActiveDialogs), after each received response, the subsequent request is sent). Therefore, no more than 'NumberOfActiveDialogs' dialogs are active simultaneously. The username and password are used for SIP Authentication when required.

The calling number of outgoing Tel to IP calls is first translated to an IP number and then (if defined) the manipulation rules are performed. The Display Name is used in the From header in addition to the IP number.

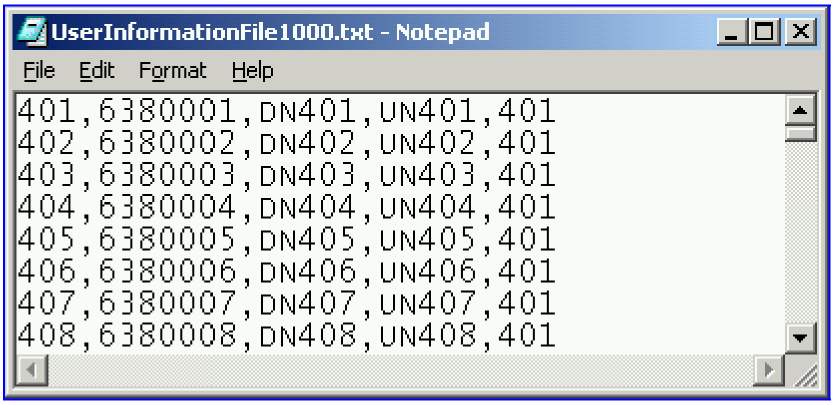
The called number of incoming IP to Tel calls is translated to a PBX extension only after manipulation rules (if defined) are performed.

The User Information file is a text file (the file size mustn't exceed 108,000 bytes) that can be loaded via the *ini* file (UserInfoFileName, described in Section 6.18 on page 201), the Embedded Web Server (refer to Section 5.8.2 on page 119) or by using the automatic update mechanism (UserInfoFileURL, refer to Section 11.3 on page 249).

Table 16-1: User Information Items

Item	Description	Maximum Size
PBX extension #	The relevant PBX extension number	10
Global IP #	The relevant IP number	20
Display name	A string that represents the PBX extensions for the Caller ID.	30
Username	A string that represents the username for SIP registration.	20
Password	A string that represents the password for SIP registration.	20

Figure 16-4: Example of a User Information File




```
401,6380001,DN401,UN401,401
402,6380002,DN402,UN402,401
403,6380003,DN403,UN403,401
404,6380004,DN404,UN404,401
405,6380005,DN405,UN405,401
406,6380006,DN406,UN406,401
407,6380007,DN407,UN407,401
408,6380008,DN408,UN408,401
```

A Selected Technical Specifications

A.1 General Specifications

The General Specifications (show in Table A-1 below) apply to the Mediant 2000, TP-1610 and TP-260 gateways. For the product-specific specifications, refer to:

- Mediant 2000 (Section A.2 on page 337).
- TP-1610 (Section A.3 on page 339).
- TP-260 (Section A.4 on page 341).

Table A-1: General Selected Technical Specifications (continues on pages 335 to 336)

Function	Specification
Voice & Tone Characteristics	
Voice Compression	G.711 PCM at 64 kbps µ-law/A-law (10, 20, 30, 40, 50, 60, 80, 100,120 msec) G.723.1 MP-MLQ at 5.3 or 6.3 kbps (30, 60, 90, 120 msec) G.726 at 32 kbps ADPCM (10, 20, 30, 40, 50, 60, 80, 100,120 msec) G.729 CS-ACELP 8 kbps Annex A / B (10, 20, 30, 40, 50, 60, 80, 100 msec) NetCoder at 6.4, 7.2, 8.0 and 8.8 kbps (20, 40, 60, 80, 100, 120 msec). EVRC (20, 40, 60, 80, 100 msec). AMR (20 msec) Transparent (20, 40, 60, 80, 100, 120 msec) GSM Full Rate (20, 40, 60, 80 msec) Microsoft GSM (40 msec) GSM-EFR (20 msec)
Silence Suppression	G.723.1 Annex A G.729 Annex B PCM and ADPCM: Standard Silence Descriptor (SID) with Proprietary Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) NetCoder
Packet Loss Concealment	G.711 appendix 1 G.723.1 G.729 a/b
Echo Cancellation	G.165 and G.168 2000, configurable tail length per gateway from 32 to 128 msec
DTMF Detection and Generation	Dynamic range 0 to -25 dBm, compliant with TIA 464B and Bellcore TR-NWT-000506.
DTMF Transport (in-band)	Mute, transfer in RTP payload or relay in compliance with RFC 2833
Call Progress Tone Detection and Generation	32 tones: single tone, dual tones or AM tones, programmable frequency & amplitude; 64 frequencies in the range 300 to 1980 Hz, 1 to 4 cadences per tone, up to 4 sets of ON/OFF periods.
Output Gain Control	-32 dB to +31 dB in steps of 1 dB
Input Gain Control	-32 dB to +31 dB in steps of 1 dB
Fax and Modem Transport Modes	
Real time Fax Relay	Group 3 real-time fax relay up to 14400 bps with auto fallback Tolerant network delay (up to 9 seconds round trip delay) T.30 (PSTN) and T.38 (IP) compliant (real-time fax) CNG tone detection & Relay per T.38 Answer tone (CED or AnsAm) detection & Relay per T.38
Fax Transparency	Automatic fax bypass (pass-through) to G.711, ADPCM or NSE bypass mode

Table A-1: General Selected Technical Specifications (continues on pages 335 to 336)

Function	Specification
Modem Transparency	Automatic switching (pass-through) to PCM, ADPCM or NSE bypass mode for modem signals (V.34 or V.90 modem detection)
Protocols	
VoIP Signaling Protocol	SIP RFC 3261
Communication Protocols	RTP/RTCP packetization. IP stack (UDP, TCP, RTP). Remote Software load (TFTP, HTTP and HTTPS).
Telephony Protocols	PRI (ETSI Euro ISDN, ANSI NI2, 4/5ESS, DMS 100, QSIG, Japan INS1500, Australian Telecom, New Zealand Telecom, Hong Kong Variant, Korean MIC) E1/T1 CAS protocols: MFC R2, E&M wink start, Immediate start, delay start, loop start, ground start, Feature Group B, D for E1/T1
In-Band Signaling	DTMF (TIA 464A) MF-R1, MFC R2 User-defined Call Progress Tones
Diagnostics	
Front panel Status LEDs	E1/T1 status LAN status Gateway status (Fail, ACT, Power, and Swap Ready).
Syslog events	Supported by Syslog Server, per RFC 3164 IETF standard.
SNMP MIBs and Traps	SNMP v2c; SNMP v3
Management	
Configuration	Gateway configuration using Web browser or *ini* files
Management and Maintenance	SNMP v2c; SNMP v3
	Syslog, per RFC 3164
	Web Management (via HTTP or HTTPS)
	Telnet

A.2 Mediant 2000 Specifications

Table A-2: Mediant 2000 Selected Technical Specifications (**continues on pages 337 to 338****)**

Function	Specification
Trunk & Channel Capacity[2]	
Capacity with E1	1, 2, 4, 8 or 16 E1 spans, 30, 60, 120, 240 or 480 digital channels: 30 Channels on 1 E1 span with gateway-1 only 60 Channels on 2 E1 spans with gateway-1 only 120 Channels on 4 E1 spans with gateway-1 only 240 Channels on 8 E1 spans with gateway-1 only 480 Channels on 16 E1 spans with gateway-1 and gateway-2
Capacity with T1	1, 2, 4, 8 or 16 T1 spans, 24, 48, 96, 192 or 384 digital channels 24 Channels on 1 T1 span with gateway-1 only 48 Channels on 2 T1 spans with gateway-1 only 96 Channels on 4 T1 spans with gateway-1 only 192 Channels on 8 T1 spans with gateway-1 only 384 Channels on 16 T1 spans with gateway-1 and gateway-2
Interfaces	
Telephony Interface	1, 2, 4, 8 or 16 E1/T1/J1 Balanced 120/100 ohm, or 75 ohm using a BNC to RJ-45 dual E1/T1 G.703 Balun adapter[3].
Network Interface	Two 10/100 Base-TX, half or full duplex with auto-negotiation
RS-232 Interface	RS-232 Terminal Interface. DB-9 connector on rear panel (available only on the 1, 2 and 4-span configuration).
LED Indicators	
LED Indications on Front Panel	Power, ACT/Fail, T1/E1 status, LAN status, Swap ready indication
Connectors & Switches	
Rear Panel	
Trunks 1 to 8 and 9 to 16	Two 50-pin female Telco connectors (DDK57AE-40500-21D) or 8 RJ-48c connectors for trunks 1 to 8 only
Ethernet 1 and 2	Two 10/100 Base-TX, RJ-45 shielded connectors
RS-232	Console port - DB-9
AC Power	Standard IEC320 Appliance inlet. Option for a dual (fully redundant) power supply.
DC Power	2-pin terminal block (screw connection type) suitable for field wiring applications connecting DC Power connector: MSTB2.5/2-STF (5.08 mm) from Phoenix Contact. Bonding and earthing: A 6-32-UNC screw is provided. Correct ring terminal and 16 AWG wire minimum must be used for connection. Or crimp connection shown below.
	Note that to meet UL approvals, users **must** fulfill the criteria below. 2-pin terminal block (crimp connection type) comprising a Phoenix Contact Adaptor: Shroud: MSTBC2,5/2-STZF-5,08. Contacts: MSTBC-MT0,5-1,0 Cable requirement: 18 AWG x 1.5 m length.

[2] Mediant 2000 channel capacity depends on configuration settings.

[3] The following Balun adaptors were tested and certified by AudioCodes:
Manufacture Name: AC&E Part Number: B04040072
Manufacture Name: RIT Part Number: R3712271

Table A-2: Mediant 2000 Selected Technical Specifications (continues on pages 337 to 338)

Function	Specification
Physical	
AC Power Supply	Universal 90 to 260 VAC 1A max, 47-63 Hz Option for a dual redundant power supply.
AC Power Consumption	1 or 2 span: 39.7 W 4 spans: 42.1 W (approximated) 8 spans: 45.3 W
DC Power Supply (optional)	36 to 72 VDC (nominal 48 VDC), 4A max, floating input
DC Power Consumption	1 or 2 span: 28.8 W 4 spans: 32.8 W 8 spans: 36.4 W
Environmental (DC)	Operation Temp: 0° to 40° C / 32° to 104° F Short Term Operation Temp (per NEBS): 0° to 55° C / 32° to 131° F Storage: -40° to 70° C / -40° to 158° F Humidity: 10 to 90% non-condensing
Environmental (AC)	Operation Temp: 0° to 40° C / 32° to 104° F Storage: -40° to 70° C / -40° to 158° F Humidity: 10 to 90% non-condensing
Hot Swap	cPCI cards are full hot swap supported Power supplies are redundant but not hot swappable
Enclosure Dimensions	445 x 44 x 300 mm; 17.5 x 1.75 x 12 inch.
Installation	1U 19-inch 2-slot cPCI chassis, Rack mount, shelf or desk top. Rack mount with 2 side brackets, option 2 extra (rear) side brackets.
Type Approvals	
Telecommunication Standards	IC CS03; FCC part 68 Chassis and Host telecom card are approved to the following telecom standards: IC CS03; FCC part 68; CTR 4, CTR 12 & CTR 13; JATE; Anatel, Mexico Telecom, Russia CCC, ASIF S016, ASIF S038.
Safety and EMC Standards	UL 60 950-1, FCC part 15 Class B, (Class A with Sun 2080 CPU card) CE Mark (EN 55022 Class B (Class A with Sun 2080 CPU card), EN 60950-1, EN 55024, EN 300 386), TS001
Environmental	NEBS Level 3: GR-63-Core, GR-1089-Core, Type 1 & 3. Approved for DC powered version. Complies with ETS 301019; ETS 300019-1, -2, -3. (T 1.1, T 2.3, T3.2). Approved for AudioCodes or DC powered versions.

A.3 TP-1610 Specifications

Table A-3: TP-1610 Selected Technical Specifications (continues on pages 339 to 340)

Function	Specification
Trunk & Channel Capacity[4]	
Capacity with E1	1, 2, 4, 8 or 16 E1 spans, 30, 60, 120, 240 or 480 digital channels 30 Channels on 1 E1 span with gateway-1 only 60 Channels on 2 E1 spans with gateway-1 only 120 Channels on 4 E1 spans with gateway-1 only 240 Channels on 8 E1 spans with gateway-1 only 480 Channels on 16 E1 spans with gateway-1 and gateway-2
Capacity with T1	1, 2, 4, 8 or 16 T1 spans, 24, 48, 96, 192 or 384 digital channels 24 Channels on 1 T1 span with gateway-1 only 48 Channels on 2 T1 spans with gateway-1 only 96 Channels on 4 T1 spans with gateway-1 only 192 Channels on 8 T1 spans with gateway-1 only 384 Channels on 16 T1 spans with gateway-1 and gateway-2
Processor	
Control Processor	Motorola PowerQUICC 8260
Control Processor Memory	SDRAM 64* - 128 MB (*on 60-channel models)
Signal Processors	AudioCodes AC486 VoIP DSP based on: TI DSP TMS5541 – each core at 133 MHz
Interfaces	
Telephony Interface	1, 2, 4, 8 or 16 E1/T1/J1 Balanced 120/100 ohm, or 75 ohm using a BNC to RJ-45 dual E1/T1 G.703 Balun adapter[5].
Network Interface	Two 10/100 Base-TX, half or full duplex with auto-negotiation
RS-232 Interface	RS-232 Terminal Interface. DB-9 connector on rear panel (available only on the 1, 2 and 4-span configuration).
PCI Bus	33 MHz, 32 bit, slave mode (PICMG 2.0 revision 2.1)
LED Indicators	
LED Indications on Front Panel	Power, ACT/Fail, T1/E1 status, LAN status, Swap ready indication
Connectors & Switches	
Rear Panel	
Trunks 1 to 8 and 9 to 16	Two 50-pin female Telco connectors (DDK57AE-40500-21D) or 8 RJ-48c connectors for trunks 1 to 8 only
Ethernet 1 and 2	Two 10/100 Base-TX, RJ-45 shielded connectors
RS-232	Console port - DB-9
Physical	
Physical	6U single cPCI slot. PICMG 2.0, R2.1 and R2.16 and R.3.0 CompactPCI™ card

[4] TP-1610 channel capacity depends on configuration settings.

[5] The following Balun adaptors were tested and certified by AudioCodes:
Manufacture Name: AC&E Part Number: B04040072
Manufacture Name: RIT Part Number: R3712271

Table A-3: TP-1610 Selected Technical Specifications (continues on pages 339 to 340)

Function	Specification
Supply Voltages and Power Consumption (typical)	480 channels 40.7 W, 3 A at 5 V, 7.8 A at 3.3 V 240 channels 24 W, 1.5 A at 5 V, 5 A at 3.3 V 120 channels 18.4 W, 0.9 A at 5 V, 4.2 A at 3.3 V
Environmental	Humidity: 10 to 90% non-condensing
Cooling	500 LFM at 50o C Ambient temp supporting 480 ports, 400 Linear Feet per Minute (LFM) at 50o C Ambient temp supporting 400 ports 300 LFM at 50o C Ambient temp supporting 240 ports
Hot Swap	Full hot swap supported boards
Type Approvals	
Telecommunication Standards	IC CS03, FCC part 68 CTR4, CTR12 , CTR13, JATE, TS.016, TSO, Anatel, Mexico Telecom
Safety and EMC Standards	UL 60950, FCC part 15 Class B, CE Mark (EN55022, EN60950, EN55024, EN300 386)

A.4 TP-260 Specifications

Table A-4: TP-260 Selected Technical Specifications

Function	Specification
Trunk & Channel Capacity [6]	
Capacity with E1	1, 2, 4 or 8 E1 spans, 30, 60, 120 or 240 digital channels
Capacity with T1	1, 2, 4 or 8 T1 spans, 24, 48, 96 or 192 digital channels
Interfaces	
PCI	Universal PCI 33/66 MHz, 32/64-bit, 3.3/5V signaling board
Telephony	Up to eight 120 Ohm-RJ-48C connectors (with adaptor)
Network	Ethernet RJ-45 connector, 10/100 Base-TX
LED Indicators	
LEDs	T1/E1 status, LAN status
Physical	
Form Factor	Full length single slot PCI card
Supply Voltages and Power Consumption (typical)	8 trunks/240 channels - 18 W (3.6 A at 5V)
Environmental	Humidity: 10 to 90% non-condensing
Cooling Requirement	500 LFM at 50^{o} C Ambient temp supporting 8 ports
Type Approvals	
Safety and EMC Standards	UL 60950, FCC part 15 Class B, CE Mark (EN55022, EN60950, EN55024)

All specifications in this document are subject to change without prior notice.

[6] TP-260 channel capacity depends on configuration settings.

Reader's Notes

B Supplied SIP Software Kit

Table B-1 describes the standard supplied software kit for the Mediant 2000, TP-1610 and TP-260 SIP gateways. The supplied documentation includes this User's Manual, the Mediant 2000 Fast Track Guide and the Mediant 3000 & Mediant 2000 & TP Series SIP Digital Release Notes.

Table B-1: Supplied Software Kit

File Name	Description
Ram.cmp file	
Mediant_SIP_xxx.cmp	Image file containing the software for the Mediant 2000 and TP-1610 gateways.
TP260_UN_SIP_xxx.cmp	Image file containing the software for the TP-260 gateway.
***ini* files and utilities**	
Mediant_SIP_T1.ini	Sample *ini* file for Mediant 2000 / TP-1610 E1 gateways.
Mediant_SIP_E1.ini	Sample *ini* file for Mediant 2000 / TP-1610 T1 gateways.
TP260_SIP_T1.ini	Sample *ini* file for TP-260 E1 gateways.
TP260_SIP_E1.ini	Sample *ini* file for TP-260 T1 gateways.
Usa_tones_xx.dat	Default loadable Call Progress Tones *dat* file.
Usa_tones_xx.ini	Call progress Tones *ini* file (used to create *dat* file).
voice_prompts.dat	Default loadable Voice Prompts *dat* file.
DConvert.exe	TrunkPack Downloadable Conversion Utility
ACSyslog08.exe	Syslog server.
bootp.exe	BootP/TFTP configuration utility
260_UNSeries.inf	A PCI driver for the TP-260 that is used to prevent the Found new Hardware Wizard to reappear each time the host PC restarts.
CAS Protocol Files	Used for various signaling types, such as *E_M_WinkTable.dat.*
MIB Files	MIB library for SNMP browser
CAS Capture Tool	Utility that is used to convert CAS traces to textual form.
ISDN Capture Tool	Utility that is used to convert ISDN traces to textual form.

Reader's Notes

C SIP Compliance Tables

The gateway complies with RFC 3261, as shown in the following sections.

C.1 SIP Functions

Table C-1: SIP Functions

Function	Supported
User Agent Client (UAC)	Yes
User Agent Server (UAS)	Yes
Proxy Server	Third-party only (Checked with Ubiquity, Delta3, Microsoft, 3Com, BroadSoft, Snom and Cisco Proxies)
Redirect Server	Third-party
Registrar Server	Third-party
Event Publication Agent (EPA)	Yes
Event State Compositore (ESC)	Third-party

C.2 SIP Methods

Table C-2: SIP Methods

Method	Supported	Comments
INVITE	Yes	
ACK	Yes	
BYE	Yes	
CANCEL	Yes	
REGISTER	Yes	Send only
REFER	Yes	
NOTIFY	Yes	
INFO	Yes	
OPTIONS	Yes	
PRACK	Yes	
UPDATE	Yes	
PUBLISH	Yes	

C.3 SIP Headers

The following SIP Headers are supported by the gateway:

Table C-3: SIP Headers (continues on pages 345 to 347)

Header Field	Supported
Accept	Yes
Accept–Encoding	Yes
Alert-Info	Yes
Allow	Yes
Also	Yes
Asserted-Identity	Yes

Table C-3: SIP Headers (continues on pages 345 to 347)

Header Field	Supported
Authorization	Yes
Call-ID	Yes
Call-Info	Yes
Contact	Yes
Content-Disposition	Yes
Content-Encoding	Yes
Content-Length	Yes
Content-Type	Yes
Cseq	Yes
Date	Yes
Diversion	Yes
Encryption	No
Expires	Yes
Fax	Yes
From	Yes
History-Info	Yes
Join	Yes
Max-Forwards	Yes
Messages-Waiting	Yes
MIN-SE	Yes
Organization	No
P-Asserted-Identity	Yes
P-Preferred-Identity	Yes
Priority	No
Proxy- Authenticate	Yes
Proxy- Authorization	Yes
Proxy- Require	Yes
Prack	Yes
Reason	Yes
Record- Route	Yes
Refer-To	Yes
Referred-By	Yes
Replaces	Yes
Require	Yes
Remote-Party-ID	Yes
Response- Key	Yes
Retry- After	Yes
Route	Yes
Rseq	Yes
Session-Expires	Yes
Server	Yes
SIP-If-Match	Yes
Subject	Yes
Supported	Yes
Timestamp	Yes
To	Yes
Unsupported	Yes
User- Agent	Yes

Table C-3: SIP Headers (continues on pages 345 to 347)

Header Field	Supported
Via	Yes
Voicemail	Yes
Warning	Yes
WWW- Authenticate	Yes

C.4 SDP Headers

The following SDP Headers are supported by the gateway:

Table C-4: SDP Headers

SDP Header Element	Supported
v - Protocol version	Yes
o - Owner/ creator and session identifier	Yes
a - Attribute information	Yes
c - Connection information	Yes
d - Digit	Yes
m - Media name and transport address	Yes
s - Session information	Yes
t - Time alive header	Yes
b - Bandwidth header	Yes
u - Uri Description Header	Yes
e - Email Address header	Yes
i - Session Info Header	Yes
p - Phone number header	Yes
y - Year	Yes

C.5 SIP Responses

The following SIP responses are supported by the gateway:

- 1xx Response - Information Responses
- 2xx Response - Successful Responses
- 3xx Response - Redirection Responses
- 4xx Response - Client Failure Responses
- 5xx Response - Server Failure Responses
- 6xx Response - Global Responses

C.5.1 1xx Response – Information Responses

Table C-5: 1xx SIP Responses

1xx Response		Supported	Comments
100	Trying	Yes	The SIP gateway generates this response upon receiving of Proceeding message from ISDN or immediately after placing a call for CAS signaling.
180	Ringing	Yes	The SIP gateway generates this response for an incoming INVITE message. On receiving this response, the gateway waits for a 200 OK response.
181	Call is being forwarded	Yes	The SIP gateway does not generate these responses. However, the gateway does receive them. The gateway processes these responses the same way that it processes the 100 Trying response.
182	Queued	Yes	The SIP gateway generates this response in Call Waiting service. When SIP gateway receives a 182 response, it plays a special waiting Ringback tone to TEL side.
183	Session Progress	Yes	The SIP gateway generates this response if Early Media feature is enabled and if the gateway plays a Ringback tone to IP

C.5.2 2xx Response – Successful Responses

Table C-6: 2xx SIP Responses

2xx Response		Supported	Comments
200	OK	Yes	
202	Accepted	Yes	

C.5.3 3xx Response – Redirection Responses

Table C-7: 3xx SIP Responses

3xx Response		Supported	Comments
300	Multiple Choice	Yes	The gateway responds with an ACK and resends the request to first in the contact list, new address.
301	Moved Permanently	Yes	The gateway responds with an ACK and resends the request to new address.
302	Moved Temporarily	Yes	The SIP gateway generates this response when call forward is used, to redirect the call to another destination. If such response is received, the calling gateway initiates an INVITE message to the new destination.
305	Use Proxy	Yes	The gateway responds with an ACK and resends the request to new address.
380	Alternate Service	Yes	"

C.5.4 4xx Response – Client Failure Responses

Table C-8: 4xx SIP Responses (continues on pages 349 to 350)

4xx Response		Supported	Comments
400	Bad Request	Yes	The gateway does not generate this response. On reception of this message, before a 200 OK has been received, the gateway responds with an ACK and disconnects the call.
401	Unauthorized	Yes	Authentication support for Basic and Digest. On receiving this message the GW issues a new request according to the scheme received on this response
402	Payment Required	Yes	The gateway does not generate this response. On reception of this message, before a 200 OK has been received, the gateway responds with an ACK and disconnects the call.
403	Forbidden	Yes	The gateway does not generate this response. On reception of this message, before a 200 OK has been received, the gateway responds with an ACK and disconnects the call.
404	Not Found	Yes	The SIP gateway generates this response if it is unable to locate the callee. On receiving this response, the gateway notifies the User with a Reorder Tone.
405	Method Not Allowed	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
406	Not Acceptable	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
407	Proxy Authentication Required	Yes	Authentication support for Basic and Digest. On receiving this message the GW issues a new request according to the scheme received on this response.
408	Request Timeout	Yes	The gateway generates this response if the no-answer timer expires. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
409	Conflict	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
410	Gone	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
411	Length Required	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
413	Request Entity Too Large	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
414	Request-URL Too Long	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
415	Unsupported Media	Yes	If the gateway receives a 415 Unsupported Media response, it notifies the User with a Reorder Tone. The gateway generates this response in case of SDP mismatch.

Table C-8: 4xx SIP Responses (continues on pages 349 to 350)

4xx Response		Supported	Comments
420	Bad Extension	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
480	Temporarily Unavailable	Yes	If the gateway receives a 480 Temporarily Unavailable response, it notifies the User with a Reorder Tone. This response is issued if there is no response from remote.
481	Call Leg/Transaction Does Not Exist	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
482	Loop Detected	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
483	Too Many Hops	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
484	Address Incomplete	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
485	Ambiguous	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.
486	Busy Here	Yes	The SIP gateway generates this response if the called party is off hook and the call cannot be presented as a call waiting call. On receiving this response, the gateway notifies the User and generates a busy tone.
487	Request Canceled	Yes	This response indicates that the initial request is terminated with a BYE or CANCEL request.
488	Not Acceptable	Yes	The gateway does not generate this response. On reception of this message, before a 200OK has been received, the gateway responds with an ACK and disconnects the call.

C.5.5 5xx Response – Server Failure Responses

Table C-9: 5xx SIP Responses

5xx Response		Comments
500	Internal Server Error	On reception of any of these Responses, the GW releases the call, sending appropriate release cause to PSTN side. The GW generates 5xx response according to PSTN release cause coming from PSTN.
501	Not Implemented	
502	Bad gateway	
503	Service Unavailable	
504	Gateway Timeout	
505	Version Not Supported	

C.5.6 6xx Response – Global Responses

Table C-10: 6xx SIP Responses

6xx Response		Comments
600	Busy Everywhere	On reception of any of these Responses, the GW releases the call, sending appropriate release cause to PSTN side.
603	Decline	
604	Does Not Exist Anywhere	
606	Not Acceptable	

Reader's Notes

D The BootP/TFTP Configuration Utility

The BootP/TFTP utility enables you to easily configure and provision our boards and media gateways. Similar to third-party BootP/TFTP utilities (which are also supported) but with added functionality; our BootP/TFTP utility can be installed on Windows™ 98 or Windows™ NT/2000/XP. The BootP/TFTP utility enables remote reset of the device to trigger the initialization procedure (BootP and TFTP). It contains BootP and TFTP utilities with specific adaptations to our requirements.

D.1 When to Use the BootP/TFTP

The BootP/TFTP utility can be used with the device as an alternative means of initializing the gateways. Initialization provides a gateway with an IP address, subnet mask, and the default gateway IP address. The tool also loads default software, *ini* and other configuration files. BootP Tool can also be used to restore a gateway to its initial configuration, such as in the following instances:

- The IP address of the gateway is not known.
- The Web browser has been inadvertently turned off.
- The Web browser password has been forgotten.
- The gateway has encountered a fault that cannot be recovered using the Web browser.

Tip: The BootP is normally used to configure the device's initial parameters. Once this information has been provided, the BootP is no longer needed. All parameters are stored in non-volatile memory and used when the BootP is not accessible.

D.2 An Overview of BootP

BootP is a protocol defined in RFC 951 and RFC 1542 that enables an internet device to discover its own IP address and the IP address of a BootP on the network, and to obtain the files from that utility that need to be loaded into the device to function.

A device that uses BootP when it powers up broadcasts a BootRequest message on the network. A BootP on the network receives this message and generates a BootReply. The BootReply indicates the IP address that should be used by the device and specifies an IP address from which the unit may load configuration files using Trivial File Transfer Protocol (TFTP) described in RFC 906 and RFC 1350.

D.3 Key Features

- Internal BootP supporting hundreds of entities.
- Internal TFTP.
- Contains all required data for our products in predefined format.
- Provides a TFTP address, enabling network separation of TFTP and BootP utilities.
- Tools to backup and restore the local database.
- Templates.
- User-defined names for each entity.
- Option for changing MAC address.
- Protection against entering faulty information.
- Remote reset.

- Unicast BootP response.
- User-initiated BootP respond, for remote provisioning over WAN.
- Filtered display of BootP requests.
- Location of other BootP utilities that contain the same MAC entity.
- Common log window for both BootP and TFTP sessions.
- Works with Windows™ 98, Windows™ NT, Windows™ 2000 and Windows™ XP.

D.4 Specifications

- BootP standards: RFC 951 and RFC 1542
- TFTP standards: RFC 1350 and RFC 906
- Operating System: Windows™ 98, Windows™ NT, Windows™ 2000 and Windows™ XP
- Max number of MAC entries: 200

D.5 Installation

➢ To install the BootP/TFTP on your computer, take these 2 steps:

1. Locate the BootP folder on the VoIP gateway supplied CD ROM and open the file Setup.exe.
2. Follow the prompts from the installation wizard to complete the installation.

➢ To open the BootP/TFTP, take these 2 steps:

1. From the **Start** menu on your computer, navigate to **Programs** and then click on **BootP**.
2. The first time that you run the BootP/TFTP, the program prompts you to set the user preferences. Refer to the Section D.10 on page 357 for information on setting the preferences.

D.6 Loading the *cmp* File, Booting the Device

Once the application is running, and the preferences were set (refer to Section D.10), for each unit that is to be supported, enter parameters into the tool to set up the network configuration information and initialization file names. Each unit is identified by a MAC address. For information on how to configure (add, delete and edit) units, refer to Section D.11 on page 359.

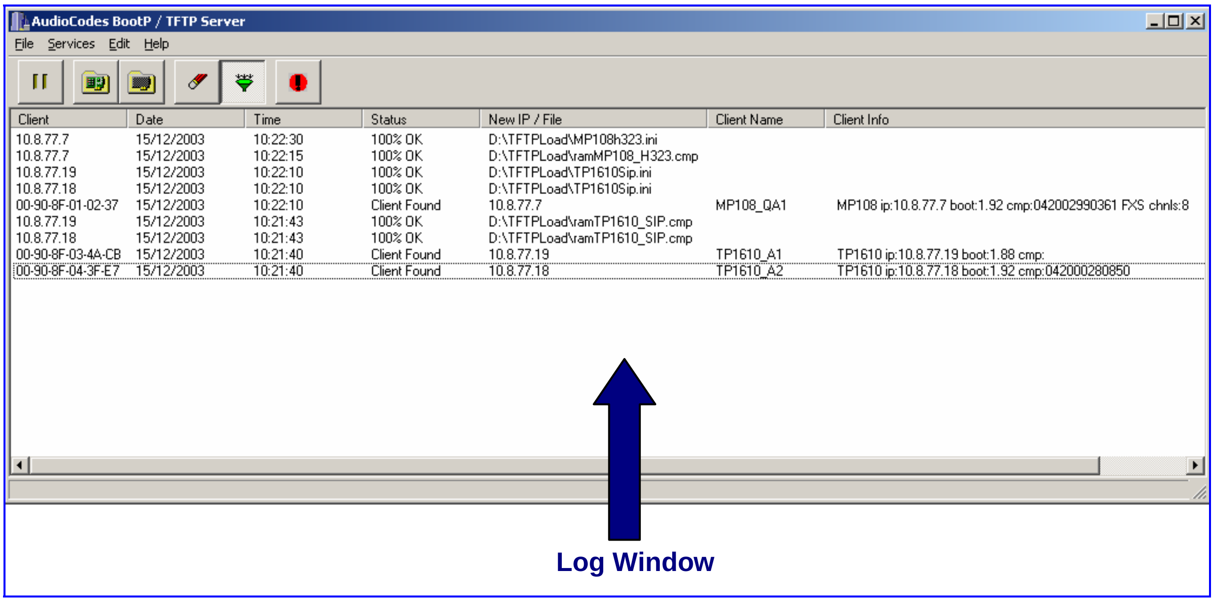
➢ To load the software and configuration files, take these 4 steps:

1. Create a folder on your computer that contains all software and configuration files that are needed as part of the TFTP process.
2. Set the BootP and TFTP preferences (refer to Section D.10).
3. Add client configuration for the VoIP gateway that you want to initialize by the BootP, refer to Section D.11.1.
4. Reset the VoIP gateway, either physically or remotely, causing the device to use BootP to access the network and configuration information.

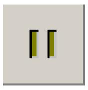
D.7 BootP/TFTP Application User Interface

Figure D-1 shows the main application screen for the BootP/TFTP utility.

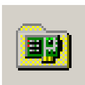
Figure D-1: Main Screen



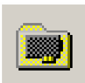
D.8 Function Buttons on the Main Screen



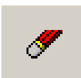
Pause: Click this button to pause the BootP Tool so that no replies are sent to BootP requests. Click the button again to restart the BootP Tool so that it responds to all BootP requests. The **Pause** button provides a depressed graphic when the feature is active.



Edit Clients: Click this button to open a new window that enables you to enter configuration information for each supported VoIP gateway. Details on the Clients window are provided in Section D.11 on page 359.

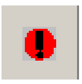


Edit Templates: Click this button to open a new window that enables you to create or edit standard templates. These templates can be used when configuring new clients that share most of the same settings. Details on the **Templates** window are provided in Section D.12 on page 364.



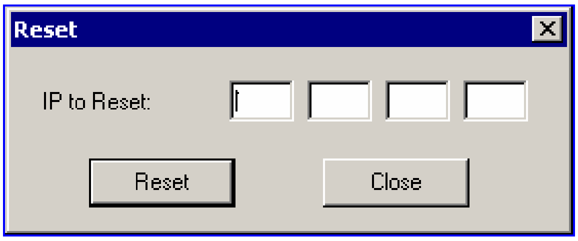
Clear Log: Click this button to clear all entries from the Log Window portion of the main application screen. Details on the log window are provided in Section D.9 on page 356.

Filter Clients: Click this button to prevent the BootP Tool from logging BootP requests received from disabled clients or from clients which do not have entries in the Clients table.



Reset: Click this button to open a new window where you enter an IP address requests for a gateway that you want to reset. Refer to Figure D-2.

Figure D-2: Reset Screen



When a gateway resets, it first sends a BootRequest. Therefore, Reset can be used to force a BootP session with a gateway without needing to power cycle the gateway. As with any BootP session, the computer running the BootP Tool must be located on the same subnet as the controlled VoIP gateway.

D.9 Log Window

The log window (refer to Figure D-1 on the previous page) records all BootP request and BootP reply transactions, as well as TFTP transactions. For each transaction, the log window displays the following information:

- **Client:** shows the Client address of the VoIP gateway, which is the MAC address of the client for BootP transactions or the IP address of the client for TFTP transactions.
- **Date:** shows the date of the transaction, based on the internal calendar of the computer.
- **Time:** shows the time of day of the transaction, based on the internal clock of the computer.
- **Status:** indicates the status of the transaction.
    - *Client Not Found:* A BootRequest was received but there is no matching client entry in the BootP Tool.
    - *Client Found:* A BootRequest was received and there is a matching client entry in the BootP Tool. A BootReply is sent.
    - *Client's MAC Changed:* There is a client entered for this IP address but with a different MAC address.
    - *Client Disabled:* A BootRequest was received and there is a matching client entry in the BootP tool but this entry is disabled.
    - *Listed At:* Another BootP utility is listed as supporting a particular client when the Test Selected Client button is clicked (for details on Testing a client, refer to Section D.11.4 on page 361).
    - *Download Status:* Progress of a TFTP load to a client, shown in %.
- **New IP / File:** shows the IP address applied to the client as a result of the BootP transaction, as well as the file name and path of a file transfer for a TFTP transaction.
- **Client Name:** shows the client name, as configured for that client in the Client Configuration screen.

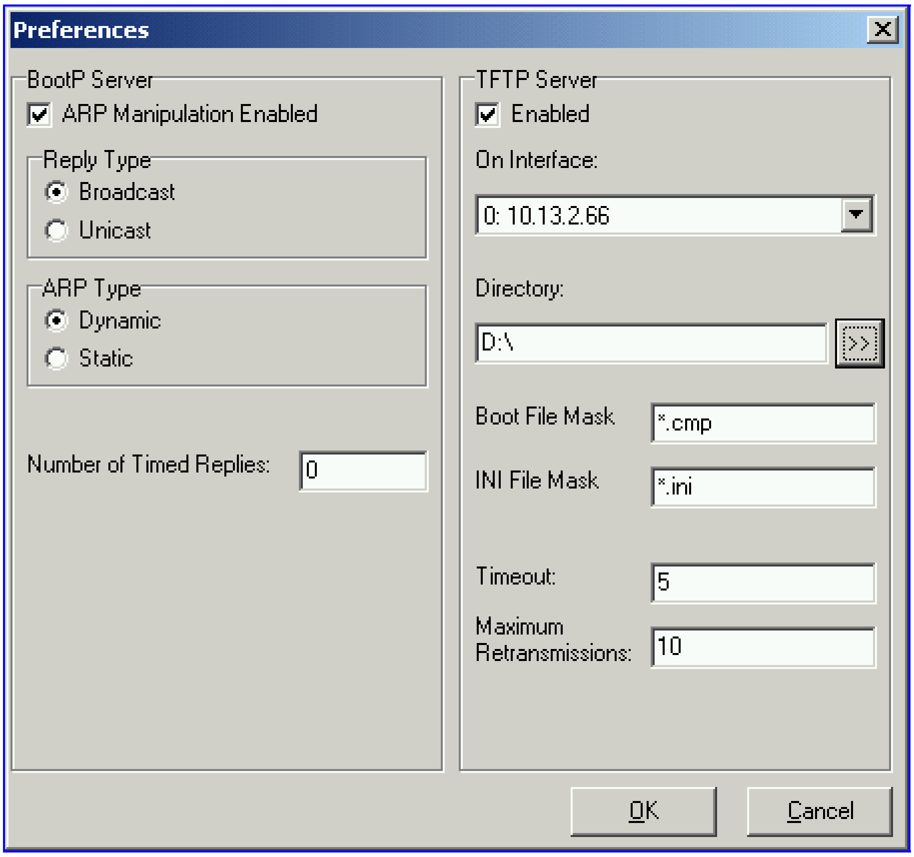
Use right-click on a line in the Log Window to open a pop-up window with the following options:

- **Reset:** Selecting this option results in a reset command being sent to the client VoIP gateway. The program searches its database for the MAC address indicated in the line. If the client is found in that database, the program adds the client MAC address to the Address Resolution Protocol (ARP) table for the computer. The program then sends a reset command to the client. This enables a reset to be sent without knowing the current IP address of the client, as long as the computer sending the reset is on the same subnet.
  Note: In order to use reset as described above, the user must have administrator privileges on the computer. Attempting to perform this type of reset without administrator privileges on the computer results in an error message. **ARP Manipulation Enable** must also be turned on in the **Preferences** window.
- **View Client:** Selecting this option, or double clicking on the line in the log window, opens the **Client Configuration** window. If the MAC address indicated on the line exists in the client database, it is highlighted. If the address is not in the client database, a new client is added with the MAC address filled out. You can enter data in the remaining fields to create a new client entry for that client.

D.10 Setting the Preferences

The Preferences window, Figure D-3, is used to configure the BootP Tool parameters.

Figure D-3: Preferences Screen



D.10.1 BootP Preferences

ARP is a common acronym for Address Resolution Protocol, and is the method used by all Internet devices to determine the link layer address, such as the Ethernet MAC address, in order to route Datagrams to devices that are on the same subnet.

When ARP Manipulation is enabled on this screen, the BootP Tool creates an ARP cache entry on your computer when it receives a BootP BootRequest from the VoIP gateway. Your computer uses this information to send messages to the VoIP gateway without using ARP again. This is particularly useful when the gateway does not yet have an IP address and, therefore, cannot respond to an ARP.

Because this feature creates an entry in the computer ARP cache, Administrator Privileges are required. If the computer is not set to allow administrator privileges, ARP Manipulation cannot be enabled.

- **ARP Manipulation Enabled:** Enable ARP Manipulation to remotely reset a gateway that does not yet have a valid IP address.

If ARP Manipulation is enabled, the following two commands are available.

- **Reply Type:** Reply to a BootRequest can be either **Broadcast** or **Unicast**. The default for the BootP Tool is **Broadcast**. In order for the reply to be set to **Unicast**, ARP Manipulation must first be enabled. This then enables the BootP Tool to find the MAC address for the client in the ARP cache so that it can send a message directly to the requesting device. Normally, this setting can be left at **Broadcast**.
- **ARP Type:** The type of entry made into the ARP cache on the computer, once **ARP Manipulation** is enabled, can be either **Dynamic** or **Static**. Dynamic entries expire after a period of time, keeping the cache clean so that stale entries do not consume computer resources. The Dynamic setting is the default setting and the setting most often used. Static entries do not expire.
- **Number of Timed Replies:** This feature is useful for communicating to VoIP gateways that are located behind a firewall that would block their BootRequest messages from getting through to the computer that is running the BootP Tool. You can set this value to any whole digit. Once set, the BootP Tool can send that number of BootReply messages to the destination immediately after you send a remote reset to a VoIP gateway at a valid IP address. This enables the replies to get through to the VoIP gateway even if the BootRequest is blocked by the firewall. To turn off this feature, set the **Number of Timed Replies** = 0.

D.10.2 TFTP Preferences

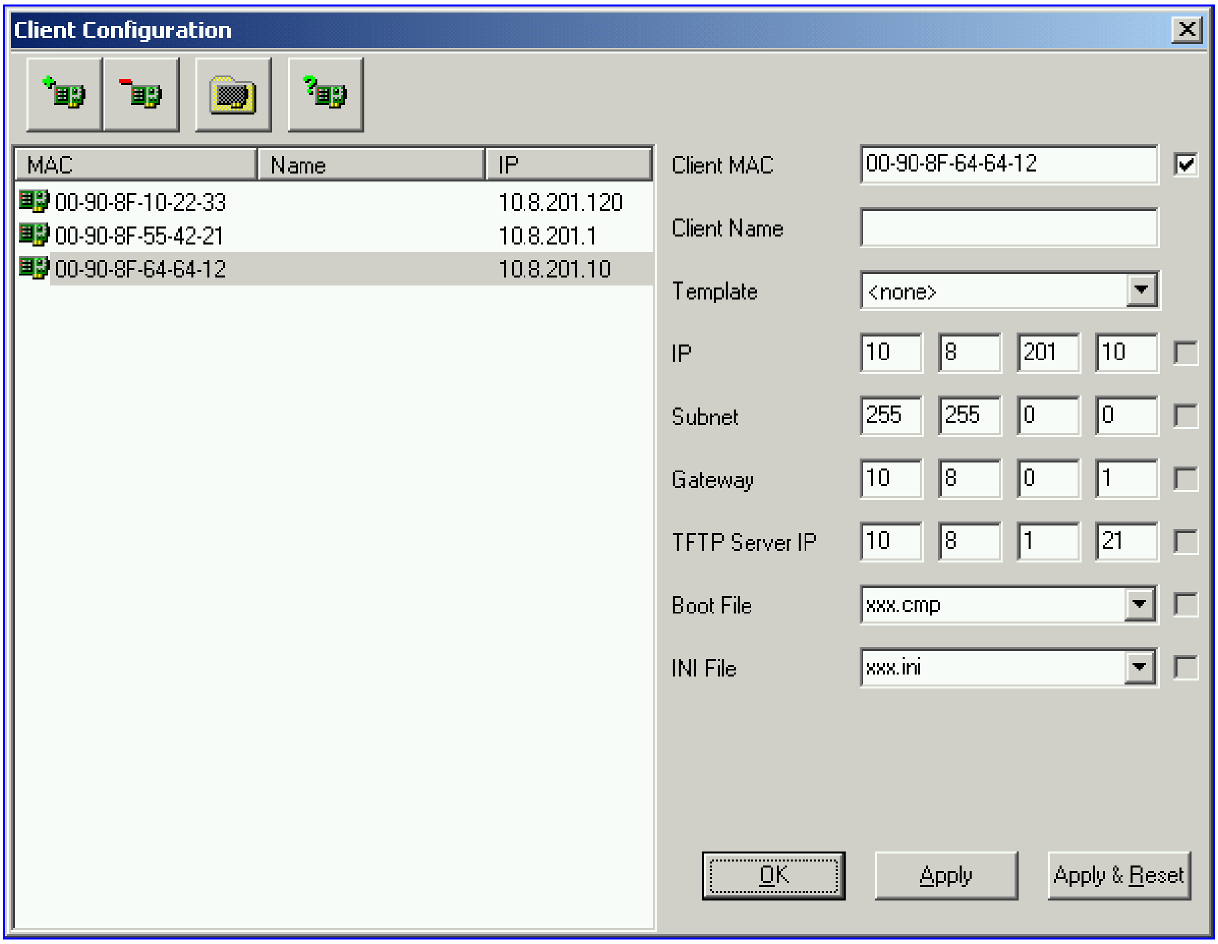
- **Enabled:** To enable the TFTP functionality of the BootP Tool, select the check box. If you want to use another TFTP application, other than the one included with the BootP Tool, unselect the box.
- **On Interface:** This pull down menu displays all network interfaces currently available on the computer. Select the interface that you want to use for the TFTP. Normally, there is only one choice.
- **Directory:** This option is enabled only when the TFTP is enabled. Use this parameter to specify the folder that contains the files for the TFTP utility to manage (*cmp*, *ini*, Call Progress Tones, etc.).
- **Boot File Mask:** Boot File Mask specifies the file extension used by the TFTP utility for the boot file that is included in the BootReply message. This is the file that contains VoIP gateway software and normally appears as *cmp*.
- ***ini* File Mask:** *ini* File mask specifies the file extension used by the TFTP utility for the configuration file that is included in the BootReply message. This is the file that contains VoIP gateway configuration parameters and normally appears as *ini*.

- **Timeout:** This specifies the number of seconds that the TFTP utility waits before retransmitting TFTP messages. This can be left at the default value of 5 (the more congested your network, the higher the value you should define in these fields).
- **Maximum Retransmissions:** This specifies the number of times that the TFTP utility tries to resend messages after timing out. This can be left at the default value of 10 (the more congested your network, the higher the value you should define in these fields).

D.11 Configuring the BootP Clients

The Clients window, shown in Figure D-4 below, is used to set up the parameters for each specific VoIP gateway.

Figure D-4: Client Configuration Screen



D.11.1 Adding Clients

Adding a client creates an entry in the BootP Tool for a specific gateway.

➢ To add a client to the list without using a template, take these 3 steps:

1. Click on the **Add New Client** icon ; a client with blank parameters is displayed.
2. Enter values in the fields on the right side of the window, using the guidelines for the fields in Section D.11.5 on page 361.
3. Click **Apply** to save this entry to the list of clients, or click **Apply & Reset** to save this entry to the list of clients and send a reset message to that gateway to immediately implement the settings.
   Note: To use **Apply & Reset** you must enable **ARP Manipulation** in the **Preferences** window. Also, you must have administrator privileges for the computer you are using.

An easy way to create several clients that use similar settings is to create a template. For information on how to create a template, refer to Section D.12 on page 364.

➢ To add a client to the list using a template, take these 5 steps:

1. Click on the **Add New Client** icon ; a client with blank parameters is displayed.
2. In the field **Template**, located on the right side of the **Client Configuration Window**, click on the down arrow to the right of the entry field and select the template that you want to use.
3. The values provided by the template are automatically entered into the parameter fields on the right side of the **Client Configuration Window**. To use the template parameters, leave the check box next to that parameter selected. The parameter values appear in gray text.
4. To change a parameter to a different value, unselect the check box to the right of that parameter. This clears the parameter provided by the template and enables you to edit the entry. Clicking the check box again restores the template settings.
5. Click **Apply** to save this entry to the list of clients or click **Apply & Reset** to save this entry to the list of clients and send a reset message to that gateway to immediately implement the settings.
   Note: To use **Apply & Reset** you must enable **ARP Manipulation** in the **Preferences** window. Also, you must have administrator privileges for the computer you are using.

D.11.2 Deleting Clients

➢ To delete a client from the BootP Tool, take these 3 steps:

1. Select the client that you wish to delete by clicking on the line in the window for that client.
2. Click the **Delete Current Client** button .
3. A warning pops up. To delete the client, click **Yes**.

D.11.3 Editing Client Parameters

➢ To edit the parameters for an existing client, take these 4 steps:

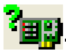
1. Select the client that you wish to edit by clicking on the line in the window for that client.
2. Parameters for that client display in the parameter fields on the right side of the window.

3. Make the changes required for each parameter.
4. Click **Apply** to save the changes, or click **Apply & Reset** to save the changes and send a reset message to that gateway to immediately implement the settings. **Note:** To use **Apply & Reset** you must enable **ARP Manipulation** in the **Preferences** window. Also, you must have administrator privileges for the computer you are using.

D.11.4 Testing the Client

There should only be one BootP utility supporting any particular client MAC active on the network at any time.

➢ To check if other BootP utilities support this client, take these 4 steps:

1. Select the client that you wish to test by clicking on the client name in the main area of the **Client Configuration Window**.
2. Click the **Test Selected Client** button .
3. Examine the Log Window on the Main Application Screen. If there is another BootP utility that supports this client MAC, there is a response indicated from that utility showing the status Listed At along with the IP address of that utility.
4. If there is another utility responding to this client, you must remove that client from either this utility or the other one.

D.11.5 Setting Client Parameters

Client parameters are listed on the right side of the **Client Configuration Window**.

- **Client MAC:** The Client MAC is used by BootP to identify the VoIP gateway. The MAC address for the VoIP gateway is printed on a label located on the VoIP gateway hardware. Enter the Ethernet MAC address for the VoIP gateway in this field. Click the box to the right of this field to enable this particular client in the BootP tool (if the client is disabled, no replies are sent to BootP requests).
  Note: When the MAC address of an existing client is edited, a new client is added, with the same parameters as the previous client.
- **Client Name:** Enter a descriptive name for this client so that it is easier to remember which VoIP gateway the record refers to. For example, this name could refer to the location of the gateway.
- **Template:** Click the pull down arrow if you wish to use one of the templates that you configured. This applies the parameters from that template to the remaining fields. Parameter values that are applied by the template are indicated by a check mark in the box to the right of that parameter. Uncheck this box if you want to enter a different value. If templates are not used, the box to the right of the parameters is colored gray and is not selectable.
- **IP:** Enter the IP address you want to apply to the VoIP gateway. Use the normal dotted decimal format.
- **Subnet:** Enter the subnet mask you want to apply to the VoIP gateway. Use the normal dotted decimal format. Ensure that the subnet mask is correct. If the address is incorrect, the VoIP gateway may not function until the entry is corrected and a BootP reset is applied.
- **Gateway:** Enter the IP address for the data network gateway used on this subnet that you want to apply to the VoIP gateway. The data network gateway is a device, such as a router, that is used in the data network to interface this subnet to the rest of the enterprise network.

- **TFTP Server IP:** This field contains the IP address of the TFTP utility that is used for file transfer of software and initialization files to the gateway. When creating a new client, this field is populated with the IP address used by the BootP Tool. If a different TFTP utility is to be used, change the IP address in this field to the IP address used by the other utility.
- **Boot File:** This field specifies the file name for the software (*cmp*) file that is loaded by the TFTP utility to the VoIP gateway after the VoIP gateway receives the BootReply message. The actual software file is located in the TFTP utility directory that is specified in the BootP **Preferences** window. The software file can be followed by command line switches. For information on available command line switches, refer to Section D.11.6 on page 362.

Notes:

- Once the software file loads into the gateway, the gateway begins functioning from that software. In order to save this software to non-volatile memory, (only the *cmp* file, i.e., the compressed firmware file, can be burned to your device's flash memory), the -fb flag must be added to the end of the file name. If the file is not saved, the gateway reverts to the old version of software after the next reset.
- The **Boot file** field can contain up to two file names: *cmp* file name to be used for load of application image and *ini* file name to be used for gateway provisioning. Either one, two or no file names can appear in the **Boot file** field. To use both file names use the ';' separator (without blank spaces) between the xxx.*cmp* and the yyy.*ini* files (e.g., *ram.cmp;SIPgw.ini*).

- ***ini* File:** This field specifies the configuration *ini* file that the gateway uses to program its various settings. Enter the name of the file that is loaded by the TFTP utility to the VoIP gateway after it receives the BootReply message. The actual *ini* file is located in the TFTP utility directory that is specified in the BootP Preferences window.

D.11.6 Using Command Line Switches

You can add command line switches in the field **Boot File**.

➢ To use a Command Line Switch, take these 4 steps:

1. In the field **Boot File**, leave the file name defined in the field as it is (e.g., ramxxx.*cmp*).
2. Place your cursor after *cmp*
3. Press the space bar
4. Type in the switch you require.

Example: 'ramxxx.*cmp* –fb' to burn flash memory.

'ramxxx.*cmp* -fb -em 4' to burn flash memory and for Ethernet Mode 4 (auto-negotiate).

Table D-1 lists and describes the switches that are available:

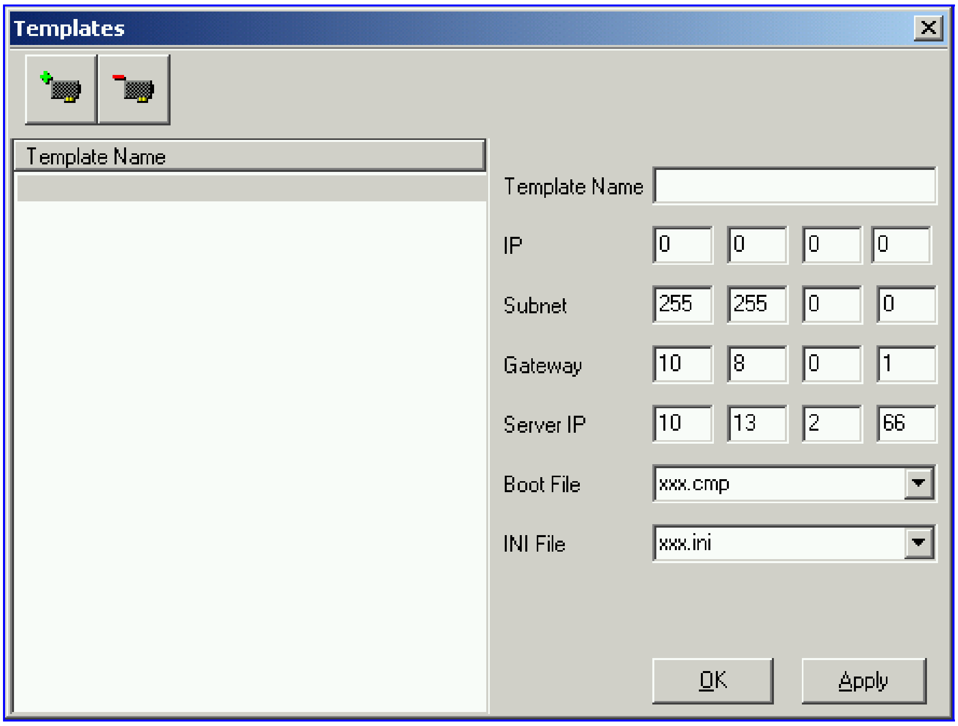
Table D-1: Command Line Switch Descriptions

<table>
<tr><th>Switch</th><th colspan="2">Description</th></tr>
<tr><td>-fb</td><td colspan="2">Burn ram.cmp in flash (only for cmp files)</td></tr>
<tr><td>-em #</td><td colspan="2">Use this switch to set Ethernet mode.
0 = 10 Base-T half-duplex
1 = 10 Base-T full-duplex
2 = 100 Base-TX half-duplex
3 = 100 Base-TX full-duplex
4 = auto-negotiate (default)
For detailed information on Ethernet interface configuration, refer to Section 9.1 on page 229.</td></tr>
<tr><td rowspan="2">-br</td><td colspan="2">This parameter is used to:
Note: This switch takes effect only from the next gateway reset.</td></tr>
<tr><td>Set the number of BootP requests the gateway sends during start-up. The gateway stops sending BootP requests when either BootP reply is received or number of retries is reached.
1 = 1 BootP retry, 1 second
2 = 2 BootP retries, 3 seconds
3 = 3 BootP retries, 6 seconds
4 = 10 BootP retries, 30 seconds
5 = 20 BootP retries, 60 seconds
6 = 40 BootP retries, 120 seconds
7 = 100 BootP retries, 300 seconds
15 = BootP retries indefinitely</td><td>Set the number of DHCP packets the gateway sends.
After all packets were sent, if there's still no reply, the gateway loads from flash.
1 = 4 DHCP packets
2 = 5 DHCP packets
3 = 6 DHCP packets (default)
4 = 7 DHCP packets
5 = 8 DHCP packets
6 = 9 DHCP packets
7 = 10 DHCP packets
15 = 18 DHCP packets</td></tr>
<tr><td>-bd</td><td colspan="2">BootP delays. Sets the interval between the device’s start-up and the first BootP/DHCP request that is issued by the device. The switch only takes effect from the next reset of the device.
1 = 1 second delay (default).
2 = 10 second delay.
3 = 30 second delay.
4 = 60 second delay.
5 = 120 second delay.</td></tr>
<tr><td>-bs</td><td colspan="2">Use –bs 1 to enable the Selective BootP mechanism.
Use –bs 0 to disable the Selective BootP mechanism.
The Selective BootP mechanism (available from Boot version 1.92) enables the gateway’s integral BootP client to filter unsolicited BootP/DHCP replies (accepts only BootP replies that contain the text ‘AUDC’ in the vendor specific information field). This option is useful in environments where enterprise BootP/DHCP servers provide undesired responses to the gateway’s BootP requests.</td></tr>
<tr><td>-be</td><td colspan="2">Use -be 1 for the device to send device-related initial startup information (such as board type, current IP address, software version) in the vendor specific information field (in the BootP request). This information can be viewed in the main screen of the BootP/TFTP, under column 'Client Info‘ (refer to Figure D-1 showing BootP/TFTP main screen with the column 'Client Info' on the extreme right). For a full list of the vendor specific Information fields, refer to Section 7.3 on page 205.
Note: This option is not available on DHCP servers.</td></tr>
</table>

D.12 Managing Client Templates

Templates can be used to simplify configuration of clients when most of the parameters are the same.

Figure D-5: Templates Screen



➢ To create a new template, take these 4 steps:

1. Click on the **Add New Template** button .
2. Fill in the default parameter values in the parameter fields.
3. Click **Apply** to save this new template.
4. Click **OK** when you are finished adding templates.

➢ To edit an existing template, take these 4 steps:

1. Select the template by clicking on its name from the list of templates in the window.
2. Make changes to the parameters, as required.
3. Click **Apply** to save this new template.
4. Click **OK** when you are finished editing templates.

➢ To delete an existing template, take these 3 steps:

1. Select the template by clicking its name from the list of templates in the window.\
2. Click on the **Delete Current Template** button .
3. A warning pop up message appears. To delete the template, click **Yes**. Note that if this template is currently in use, the template cannot be deleted.

E RTP/RTCP Payload Types and Port Allocation

RTP Payload Types are defined in RFC 3550 and RFC 3551. We have added new payload types to enable advanced use of other coder types. These types are reportedly not used by other applications.

E.1 Payload Types Defined in RFC 3551

Table E-1: Packet Types Defined in RFC 3551

Payload Type	Description	Basic Packet Rate [msec]
0	G.711 µ-Law	10,20
2	G.726-32	10,20
3	GSM-FR	20
4	G.723 (6.3/5.3 kbps)	30
8	G.711 A-Law	10,20
18	G.729A/B	20
200	RTCP Sender Report	Randomly, approximately every 5 seconds (when packets are sent by channel)
201	RTCP Receiver Report	Randomly, approximately every 5 seconds (when channel is only receiving)
202	RTCP SDES packet	
203	RTCP BYE packet	
204	RTCP APP packet	

E.2 Defined Payload Types

Table E-2: Defined Payload Types

Payload Type	Description	Basic Packet Rate [msec]
3	MS-GSM	40
3	GSM-EFR	20
51	NetCoder 6.4 kbps	20
52	NetCoder 7.2 kbps	20
53	NetCoder 8.0 kbps	20
54	NetCoder 8.8 kbps	20
56	Transparent PCM	20
60	EVRC	20
64	AMR	20
96	DTMF relay per RFC 2833	
102	Fax Bypass	20
103	Modem Bypass	20
104	RFC 2198 (Redundancy)	Same as channel's voice coder.
105	NSE Bypass	

E.3 Default RTP/RTCP/T.38 Port Allocation

The following table describes gateway's default RTP/RTCP/T.38 port allocation.

Table E-3: Default RTP/RTCP/T.38 Port Allocation

Channel Number	RTP Port	RTCP Port	T.38 Port
1	6000	6001	6002
2	6010	6011	6012
3	6020	6021	6022
4	6030	6031	6032
5	6040	6041	6042
6	6050	6051	6052
7	6060	6061	6062
8	6070	6071	6072
:	:	:	:
n	**6000 + 10(n-1)**	**6001 + 10(n-1)**	**6002 + 10(n-1)**
:	:	:	:
96	6950	6951	6952
:	:	:	:
120	7190	7191	7192
:	:	:	:
192	7910	7911	7912
:	:	:	:
240	8390	8391	8392
:	:	:	:
384	9830	9831	9832
:	:	:	:
480	10790	10791	10792

Notes:

- To configure the gateway to use the same port for both RTP and T.38 packets, set the parameter 'T38UseRTPPort' to 1.
- The number of cahnnels depends on the gateway (i.e., Mediant 2000 with one or two TP-1610 boards, or TP-260).

F RTP Control Protocol Extended Reports (RTCP-XR)

RTP Control Protocol Extended Reports (RTCP-XR) is a VoIP management control that defines a set of metrics containing information for assessing VoIP call quality and diagnosing problems. RTCP-XR (RFC 3611) extends the RTCP reports defined in RFC 3550 by providing additional VoIP metrics.

RTCP-XR information publishing is implemented in the media server according to <draft-johnston-sipping-rtcp-summary-07>. This draft defines how a SIP User Agent (UA) publishes the detailed information to a defined collector.

RTCP-XR messages containing key call-quality-related metrics are exchanged periodically (user-defined) between the gateway and the SIP UA. This allows an analyzer to monitor these metrics midstream, or a gateway to retrieve them using SNMP. The gateway can send RTCP-XR reports to an Event State Compositor (ESC) server using PUBLISH messages. These reports can be sent at the end of each call (configured using RTCPXRReportMode) and according to a user-defined interval (RTCPInterval or DisableRTCPRandomize) between consecutive reports.

To enable RTCP-XR reporting, the VQMonEnable *ini* file parameter must be set to 1.

For a detailed description of the RTCP-XR ini file parameters, refer to Table 6-12 on page 196.

RTCP-XR measures VoIP call quality such as packet loss, delay, signal / noise / echo levels, estimated R-factor, and mean opinion score (MOS). RTCP-XR measures these parameters using the metrics listed in the table below.

Table F-1: RTCP-XR Published VoIP Metrics (continues on pages 367 to 368)

Metric Name
General
Start Timestamp
Stop Timestamp
Call-ID
Local Address (IP, Port & SSRC)
Remote Address (IP, Port & SSRC)
Session Description
Payload Type
Payload Description
Sample Rate
Frame Duration
Frame Octets
Frames per Packets
Packet Loss Concealment
Silence Suppression State
Jitter Buffer
Jitter Buffer Adaptive
Jitter Buffer Rate
Jitter Buffer Nominal
Jitter Buffer Max
Jitter Buffer Abs Max
Packet Loss
Network Packet Loss Rate
Jitter Buffer Discard Rate

Table F-1: RTCP-XR Published VoIP Metrics (continues on pages 367 to 368)

Metric Name
Burst Gap Loss
Burst Loss Density
Burst Duration
Gap Loss Density
Gap Duration
Minimum Gap Threshold
Delay
Round Trip Delay
End System Delay
One Way Delay
Interarrival Jitter
Min Absolute Jitter
Signal
Signal Level
Noise Level
Residual Echo Return Noise
Quality Estimates
Listening Quality R
RLQ Est. Algorithm
Conversational Quality R
RCQ Est. Algorithm
External R In
Ext. R In Est. Algorithm
External R Out
Ext. R Out Est. Algorithm
MOS-LQ
MOS-LQ Est. Algorithm
MOS-CQ
MOS-CQ Est. Algorithm
QoE Est. Algorithm

G Accessory Programs and Tools

The accessory applications and tools shipped with the device provide you with friendly interfaces that enhance device usability and smooth your transition to the new VoIP infrastructure. The following applications are available:

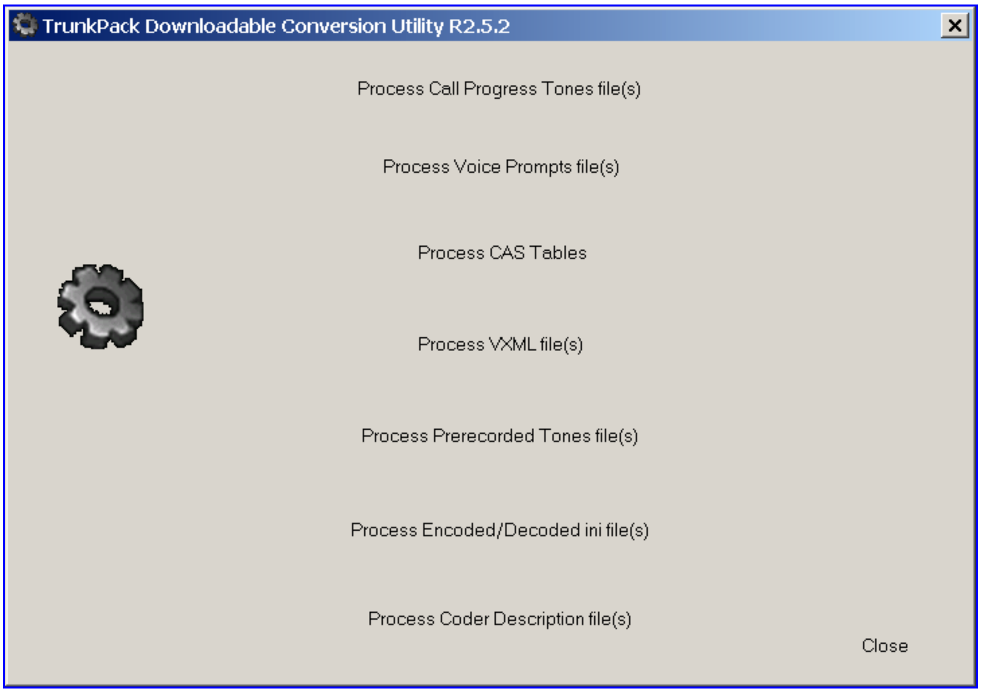
- TrunkPack Downloadable Conversion Utility (refer to Section G.1 below).
- PSTN Trace Utility (refer to Section G.2 on page 376).

G.1 TrunkPack Downloadable Conversion Utility

Use the TrunkPack Downloadable Conversion Utility to:

- Create a loadable Call Progress Tones file (refer to Section G.1.1 on page 370).
- Create a loadable Voice Prompts file from pre-recorded voice messages (refer to Section G.1.2 on page 371).
- To create a loadable CAS protocol table file (refer to Section G.1.3 on page 372).
- Encode / decode an *ini* file (refer to Section G.1.4 on page 374).
- Create a loadable Prerecorded Tones file (refer to Section G.1.5 on page 375).

Figure G-1: TrunkPack Downloadable Conversion Utility Opening Screen



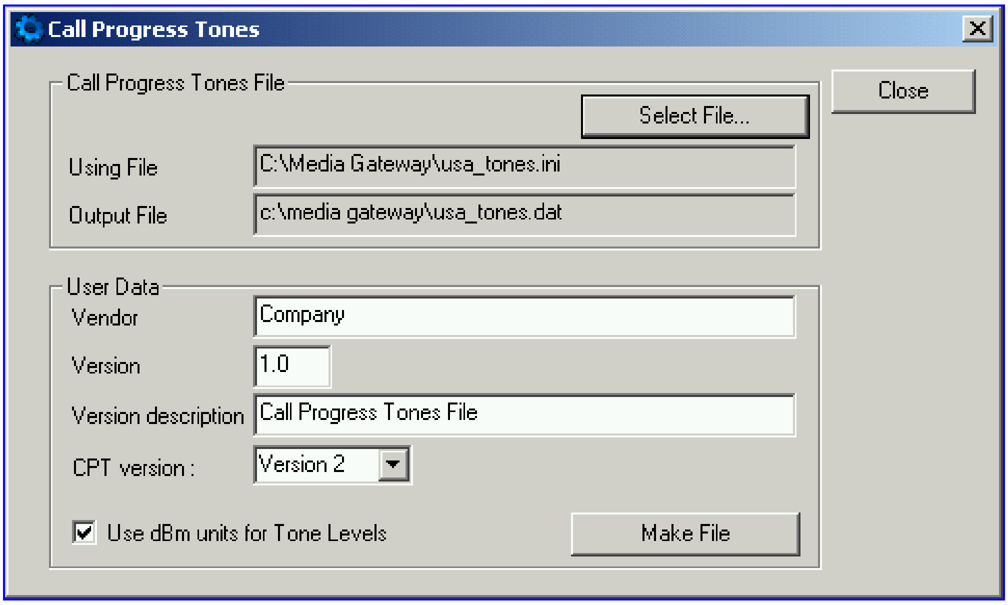
G.1.1 Converting a CPT *ini* File to a Binary *dat* File

For detailed information on creating a CPT *ini* file, refer to Section 16.1 on page 329.

➢ To convert a CPT *ini* file to a binary *dat* file, take these 10 steps:

1. Execute the TrunkPack Downloadable Conversion Utility, DConvert.exe (supplied with the software package); the utility's main screen opens (shown in Figure G-1).
2. Click the **Process Call Progress Tones File(s)** button; the Call Progress Tones screen, shown in Figure G-2, opens.

Figure G-2: Call Progress Tones Conversion Screen



3. Click the **Select File…** button that is in the 'Call Progress Tone File' box.
4. Navigate to the folder that contains the CPT *ini* file you want to convert.
5. Click the *ini* file and click the **Open** button; the name and path of both the *ini* file and the (output) *dat* file appears in the fields below the Select File button.
6. Enter the Vendor Name, Version Number and Version Description in the corresponding required fields under the 'User Data' section.
   - The maximum length of the Vendor field is 256 characters.
   - The format of the Version field is composed of two integers separated by a period '.' (e.g., 1.2, 23.4, 5.22).
   - The maximum length of the Version Description field is 256 characters.
7. The default value of the CPT Version drop-down list is Version 3. Do one of the following:
   - If the software version you are using is prior to version 4.4, select Version 1 (to maintain backward compatibility).
   - If the software version you are using is 4.4, select Version 2.
   - Otherwise, leave the value at its default.
8. Check the 'Use dBm units for Tone Levels' check box. Note that the levels of the Call Progress Tones (in the CPT file) must be in -dBm units.
9. Click the **Make File** button; you're prompted that the operation (conversion) was successful.
10. Close the application.

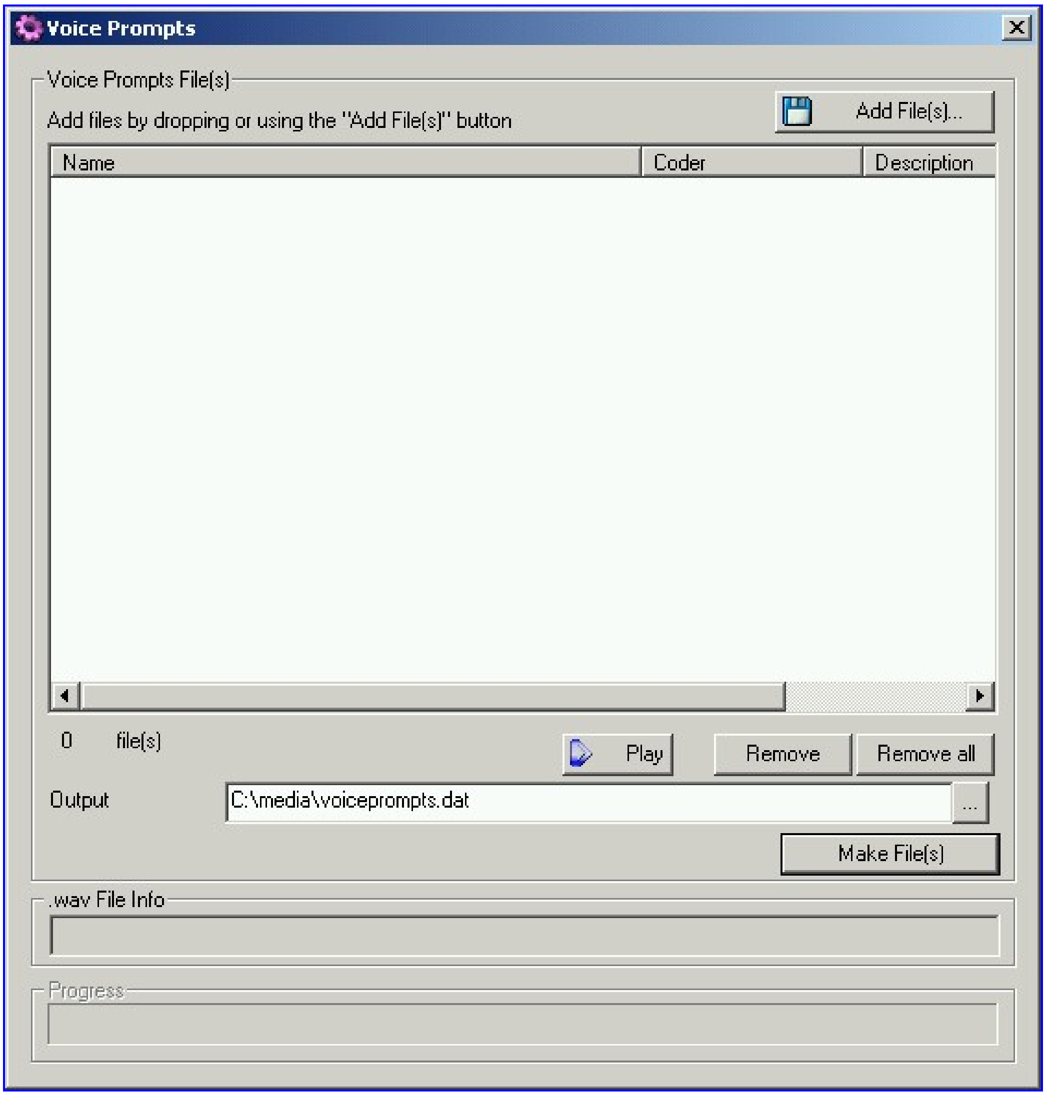
G.1.2 Creating a Loadable Voice Prompts File

For detailed information on the Voice Prompts file, refer to Section 16.3 on page 331.

➢ **To create a loadable Voice Prompts *dat* file from your voice recording files, take these 7 steps:**

1. Execute the TrunkPack Downloadable Conversion Utility, DConvert.exe (supplied with the software package); the utility's main screen opens (shown in Figure G-1).
2. Click the **Process Voice Prompts File(s)** button; the Voice Prompts screen, shown in Figure G-3, opens.

Figure G-3: Voice Prompts Screen



3. To add the pre-recorded voice files to the 'Voice Prompts' screen follow one of these procedures:
   - Select the files and drag them to the 'Voice Prompts' screen.
   - Click the **Add File(s)** button; the 'Select Files' screen opens. Select the required Voice Prompt files and click the **Add>>** button. Close the 'Select Files' screen.

4. Arrange the files according to your requirements by dragging and dropping them from one location in the list to another. Note that the sequence of the files determines their assigned Voice Prompt ID.

> **Tip 1:** Use the **Play** button to play *wav* files through your PC speakers.
>
> **Tip 2:** Use the **Remove** and **Remove all** buttons to delete files from the list.

5. For each of the raw files, select a coder that corresponds with the coder it was *originally* recorded in by completing the following steps:
   - Double-click or right-click the required file(s); the 'File Data' window (shown in Figure G-4) appears.
   - From the 'Coder' drop-down list, select the required coder type.
   - In the 'Description' field, enter additional identifying information.
   - Close the 'File Data' window.
   - Note that for *wav* files, a coder is automatically selected from the *wav* file's header.

Figure G-4: File Data Window

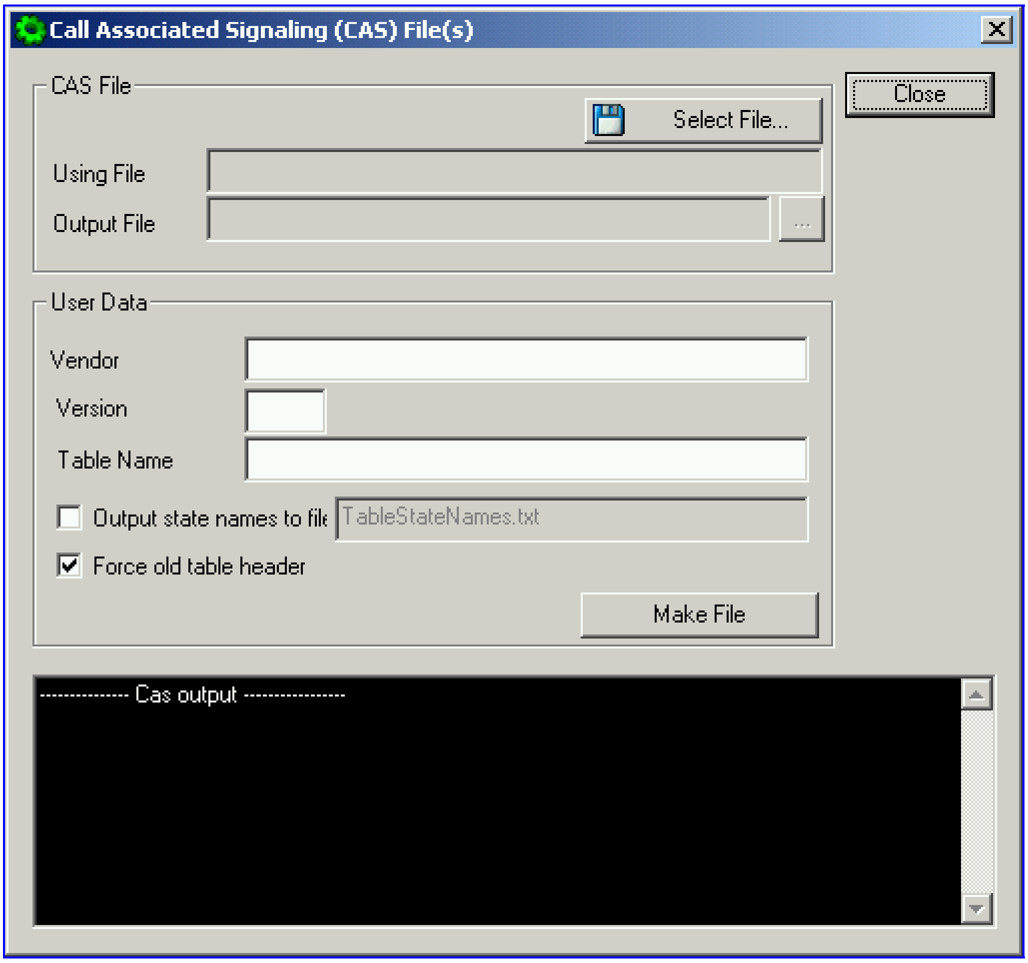
6. In the 'Output' field, specify the output directory in which the Voice Prompts file is generated followed by the name of the Voice Prompts file (the default name is *voiceprompts.dat*).
7. Click the **Make File(s)** button; the Voice Prompts loadable file is produced.

G.1.3 Creating a loadable CAS Protocol Table File

➢ To create a loadable CAS protocol table file, take these 11 steps:

1. Construct the CAS protocol files (xxx.txt and UserProt_defines_xxx.h).
2. Copy the files generated in the previous step to the same directory the TrunkPack Downloadable Conversion utility is located and ensure that the files CASSetup.h and cpp.exe are also located in the same directory.
3. Execute the TrunkPack Downloadable Conversion utility, DConvert.exe (supplied with the software package); the utility's main screen opens (shown in Figure G-1).
4. Click the button **Process CAS Tables**; the Call Associated Signaling (CAS) screen, shown in Figure G-5 below, opens.

Figure G-5: Call Associated Signaling (CAS) Screen



5. Click the button **Select File…** under the section ‘CAS File’; a Browse window appears.
6. Navigate to the desired location and select the *txt* file you want to converted; this automatically designates the output file as the same name and path, but with a *dat* extension, the table’s name is also automatically designated.
7. Enter the vendor name and version number in the required fields under the section ‘User Data’:
   - **Vendor** - maximum of 32 characters.
   - **Version** - must be in the format: [number] [single period ‘.’] [number] (e.g., 1.2, 23.4, 5.22).
8. Modify the **Table Name** according to your requirements.
9. To create a file (for troubleshooting purposes) that contains the name of the States and their actual values:

   Check the 'Output state names to file' checkbox; the default file name *TableStateNames.txt* appears in the adjacent field (you can modify the name of the file). The generated file is to be located in the same directory as the TrunkPack Downloadable Conversion utility.
10. Uncheck the **Force old table header** checkbox.
11. Click the button **Make File'**; the *dat* file is generated and placed in the directory specified in the field 'Output File. A message box informing you that the operation was successful indicates that the process is completed.
    On the bottom of the Call Assisted Signaling (CAS) Files(s) screen, the CAS output log box displays the log generated by the process. It can be copied as needed. The information in it isn’t retained after the screen is closed.

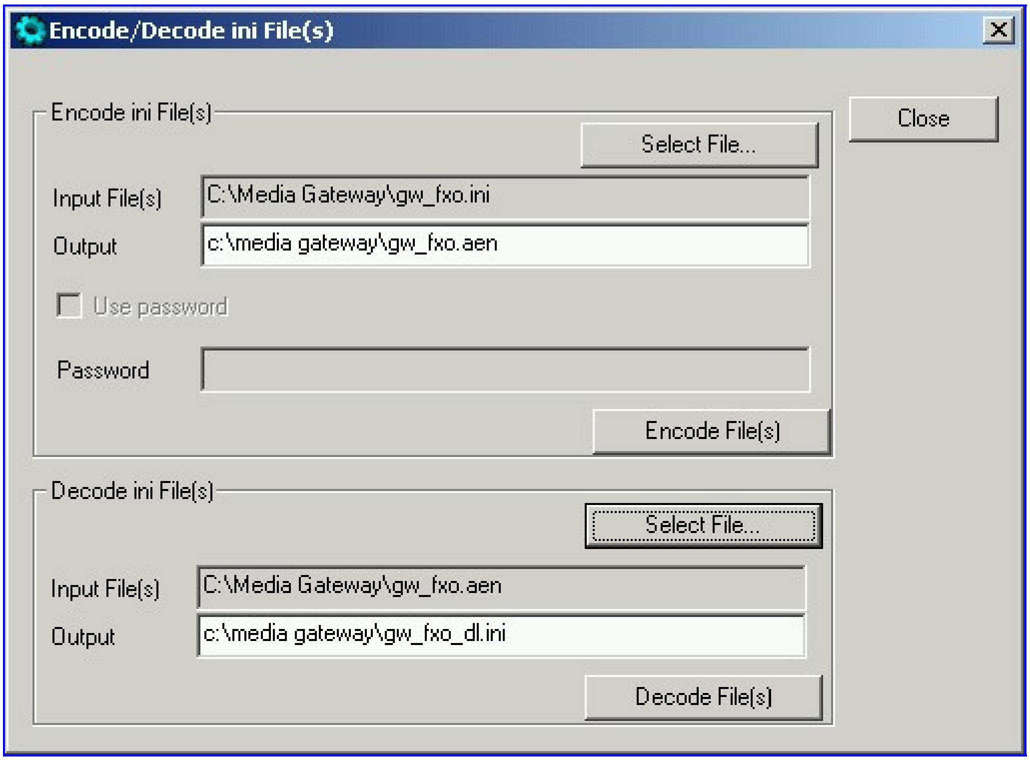
G.1.4 Encoding / Decoding an *ini* File

For detailed information on secured *ini* file, refer to Section 6.1 on page 127.

➢ To encode an *ini* file, take these 6 steps:

1. Execute the TrunkPack Downloadable Conversion Utility, DConvert.exe (supplied with the software package); the utility's main screen opens (shown in Figure G-1).
2. Click the **Process Encoded/Decoded *ini* file(s)** button; the 'Encode/Decode *ini* File(s)' screen, shown in Figure G-6, opens.

Figure G-6: Encode/Decode *ini* File(s) Screen



3. Click the **Select File…** button under the 'Encode *ini* File(s)' section.
4. Navigate to the folder that contains the *ini* file you want to encode.
5. Click the *ini* file, and then click the **Open** button; the name and path of both the *ini* file and the output encoded file appear in the fields under the **Select File** button. Note that the name and extension of the output file can be modified.
6. Click the **Encode File(s)** button; an encoded *ini* file with the name and extension you specified is created.

➢ To decode an encoded *ini* file, take these 4 steps:

1. Click the **Select File…** button under the 'Decode *ini* File(s)' section.
2. Navigate to the folder that contains the file you want to decode.
3. Click the file and click the **Open** button. the name and path of both the encode *ini* file and the output decoded file appear in the fields under the **Select File** button. Note that the name of the output file can be modified.
4. Click the **Decode File(s)** button; a decoded *ini* file with the name you specified is created.

Note that the decoding process verifies the input file for validity. Any change made to the encoded file causes an error and the decoding process is aborted.

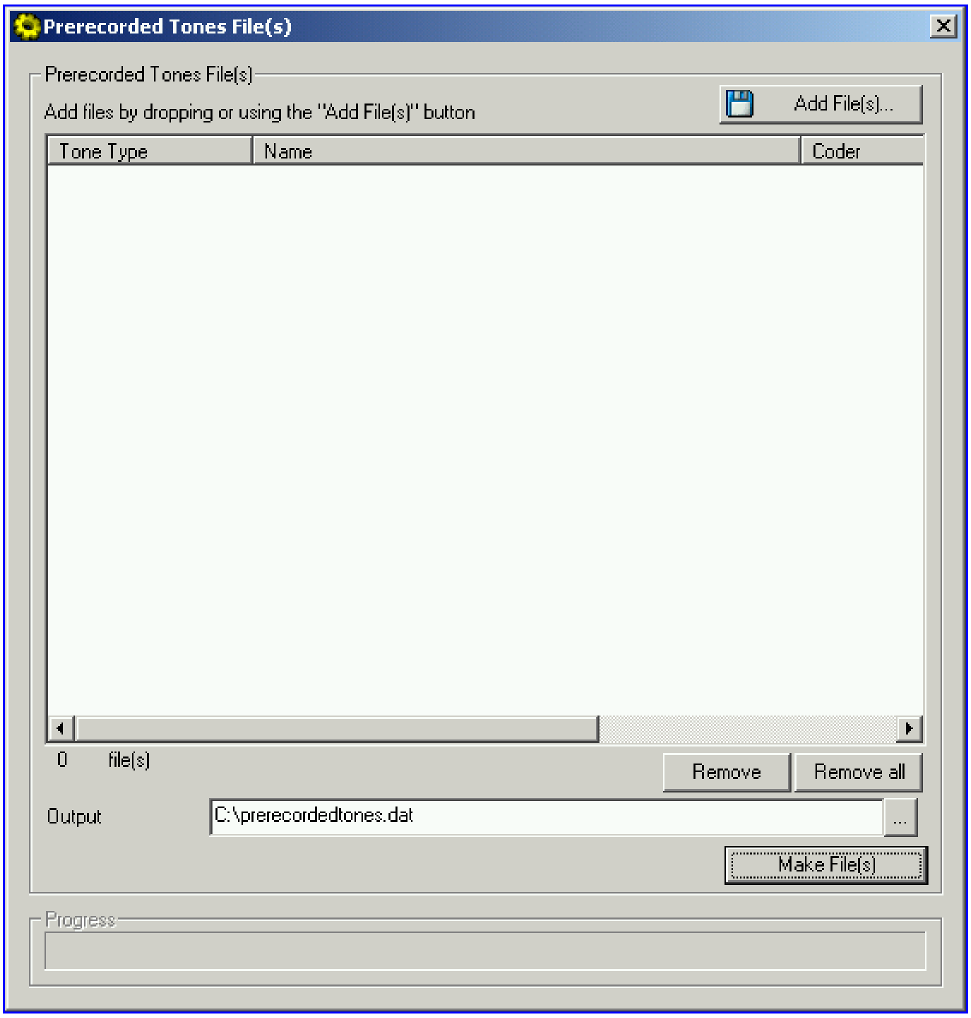
G.1.5 Creating a Loadable Prerecorded Tones File

For detailed information on the PRT file, refer to Section 16.2.1 on page 332.

➢ **To create a loadable PRT *dat* file from your raw data files, take these 7 steps:**

1. Prepare the prerecorded tones (raw data PCM or L8) files you want to combine into a single *dat* file using standard recording utilities.
2. Execute the TrunkPack Downloadable Conversion utility, DConvert.exe (supplied with the software package); the utility's main screen opens (shown in Figure G-1).
3. Click the **Process Prerecorded Tones File(s)** button; the Prerecorded Tones File(s) screen, shown in Figure G-3, opens.

Figure G-7: Prerecorded Tones Screen



4. To add the prerecorded tone files (you created in Step 1) to the 'Prerecorded Tones' screen follow one of these procedures:
   - Select the files and drag them to the 'Prerecorded Tones' screen.
   - Click the **Add File(s)** button; the 'Select Files' screen opens. Select the required Prerecorded Tone files and click the **Add>>** button. Close the 'Select Files' screen.

5. For each raw data file, define a Tone Type, a Coder and a Default Duration by completing the following 6 steps:
   - Double-click or right-click the required file; the 'File Data' window (shown in Figure G-4) appears.
   - From the 'Type' drop-down list, select the tone type this raw data file is associated with.
   - From the 'Coder' drop-down list, select the coder that corresponds to the coder this raw data file was *originally* recorded with.
   - In the 'Description' field, enter additional identifying information (optional).
   - In the 'Default' field, enter the default duration this raw data file is repeatedly played.
   - Close the 'File Data' window (press the **Esc** key to cancel your changes); you are returned to the Prerecorded Tones File(s) screen.

Figure G-8: File Data Window

6. In the 'Output' field, specify the output directory in which the PRT file is generated followed by the name of the PRT file (the default name is *prerecordedtones.dat*). Alternatively, use the Browse button to select a different output file. Navigate to the desired file and select it; the selected file name and its path appear in the 'Output' field.
7. Click the **Make File(s)** button; the Progress bar at the bottom of the window is activated. The *dat* file is generated and placed in the directory specified in the 'Output' field. A message box informing you that the operation was successful indicates that the process is completed.

G.2 PSTN Trace Utility

These utilities are designed to convert PSTN trace binary files to textual form. The binary PSTN trace files are generated when the user sets the PSTN interface to trace mode.

G.2.1 Operation

- Generating textual trace/audit file for CAS protocols:

  To generate a readable text file out of the binary trace file when using CAS protocols, rename the PSTN trace binary file to CASTrace0.dat and copy it to the same directory in which the translation utility CAS_Trace.exe is located. Then, run CAS_Trace.exe (no arguments are required). As a result, the textual file CASTrace0.txt is created.

- Generating textual trace/audit file for ISDN PRI protocols:

  To generate a readable text file out of the binary trace file when using ISDN protocols, copy the PSTN trace binary file to the same directory in which the translation utility **Convert_Trace.bat** is located. The following files should reside in the same directory: **Dumpview.exe, Dumpview.cfg** and **ReadMe.txt**. Please read carefully the **ReadMe.txt** in order to understand the usage of the translation utility. Next, run the **Convert_Trace.bat**. As a result, the textual file is created.

To start and collect the PSTN trace via the Web, please use the following instructions. (Refer to Figure G-9 for a view of the Trunk Traces). Also, please note if the PSTN trace was of a PRI of CAS collection based on the framer involved in the trace. This information is needed to properly parse the captured data.

➢ To start and collect the PSTN trace via the Web, take these 10 steps:

1. Run the UDP2File utility.
2. Determine the trace file name.
3. Determine the UDP port.
4. Mark the 'PSTN Trace' check box.
5. Click the **Run** button; the UDP2File utility starts to collect the trace messages.
6. Activate the Web page by entering <gateway'S IP address>/TrunkTraces, such as: http://10.8.8.101/TrunkTraces. The user and password is the same for the unit.
7. In the Web page set the trace level of each trunk.
8. Enable the trace via the Web.
9. Determine the UDP port (the same as in step 3).
10. Click the **Submit** button; the board starts to send the trace messages. In the UDP2File utility (Refer to Figure G-10) you should see the number in the packets counter increasing.

Figure G-9: Trunk Traces

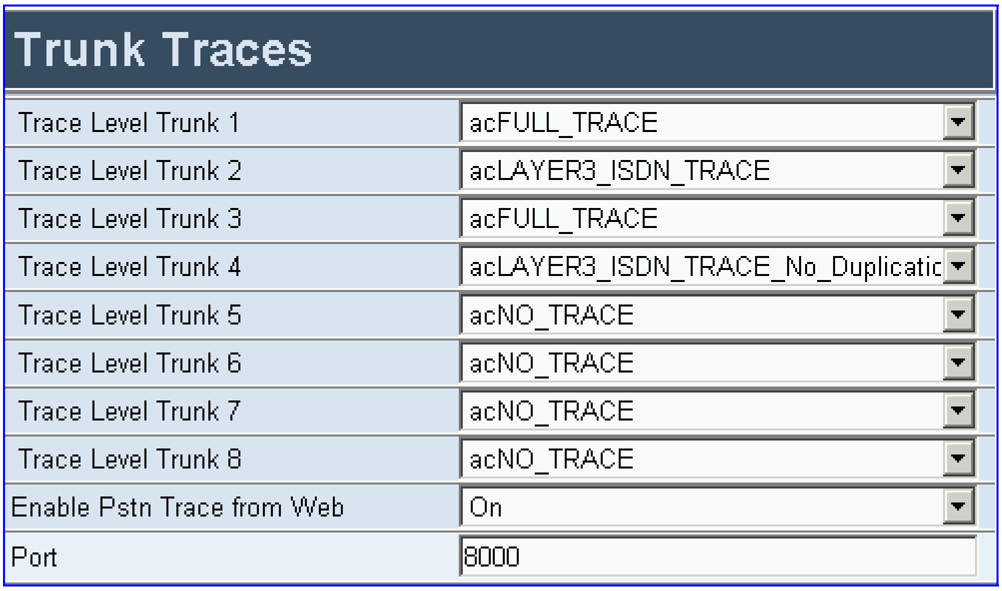
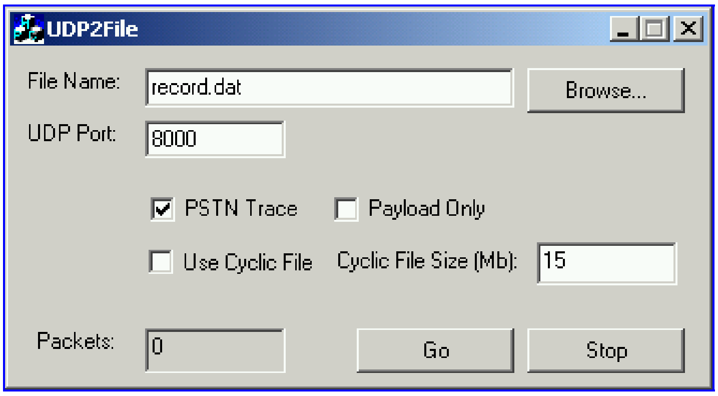


Figure G-10: UDP2File Utility



Reader's Notes

H Release Reason Mapping

This appendix describes the available mapping mechanisms of SIP Responses to Q.850 Release Causes and vice versa.

Table H-1 and Table H-2 describe the existing mapping of ISDN Release Causes to SIP Responses. To override this hard-coded mapping and flexibly map SIP Responses to ISDN Release Causes use the parameters CauseMapISDN2SIP and CauseMapSIP2ISDN (described in Section 6.14 on page 172), or via the Embedded Web Server (refer to Section 5.5.5.6 on page 77).

It is also possible to map the less commonly-used SIP Responses to a single default ISDN Release Cause. Use the parameter DefaultCauseMapISDN2IP (described in Section 6.14 on page 172) to define a default ISDN Cause that is always used except when the following Release Causes are received: Normal Call Clearing (16), User Busy (17), No User Responding (18) or No Answer from User (19). This mechanism is only available for Tel to IP calls.

H.1 Reason Header

The gateway supports the Reason header according to RFC 3326. The Reason header is used to convey information describing the disconnection cause of a call:

- **Sending Reason header:** If a call is disconnected from the Tel side (ISDN), the Reason header is set to the received Q.850 cause in the appropriate message (BYE / CANCEL / final failure response) and sent to the SIP side. If the call is disconnected because of a SIP reason, the Reason header is set to the appropriate SIP response.
- **Receiving Reason header:** If a call is disconnected from the IP side and the SIP message includes the Reason header, it is sent to the Tel side according to the following logic:
    - If the Reason header includes a Q.850 cause, it is sent as is.
    - If the Reason header includes a SIP response: (1) If the message is a final response, the response status code is translated to Q.850 format and passed to ISDN. (2) If the message isn’t a final response, it is translated to a Q.850 cause.
    - When the Reason header is received twice (i.e., SIP Reason and Q.850), the Q.850 takes precedence over the SIP reason and is sent to the Tel side.

H.2 Fixed Mapping of ISDN Release Reason to SIP Response

Table H-1 below describes the mapping of ISDN release reason to SIP response.

Table H-1: Mapping of ISDN Release Reason to SIP Response
(continues on pages 380 to 381)

ISDN Release Reason	Description	SIP Response	Description
1	Unallocated number	404	Not found
2	No route to network	404	Not found
3	No route to destination	404	Not found
6	Channel unacceptable	406*	Not acceptable
7	Call awarded and being delivered in an established channel	500	Server internal error
16	Normal call clearing	-*	BYE
17	User busy	486	Busy here
18	No user responding	408	Request timeout
19	No answer from the user	480	Temporarily unavailable
20	Subscriber absent	480	Temporarily unavailable
21	Call rejected	403	Forbidden
22	Number changed w/o diagnostic	410	Gone
22	Number changed with diagnostic	410	Gone
23	Redirection to new destination	480	Temporarily unavailable
26	Non-selected user clearing	404	Not found
27	Destination out of order	502	Bad gateway
28	Address incomplete	484	Address incomplete
29	Facility rejected	501	Not implemented
30	Response to status enquiry	501*	Not implemented
31	Normal unspecified	480	Temporarily unavailable
34	No circuit available	503	Service unavailable
38	Network out of order	503	Service unavailable
41	Temporary failure	503	Service unavailable
42	Switching equipment congestion	503	Service unavailable
43	Access information discarded	502*	Bad gateway
44	Requested channel not available	503*	Service unavailable
47	Resource unavailable	503	Service unavailable
49	QoS unavailable	503*	Service unavailable
50	Facility not subscribed	503*	Service unavailable
55	Incoming calls barred within CUG	403	Forbidden
57	Bearer capability not authorized	403	Forbidden
58	Bearer capability not presently available	503	Service unavailable
63	Service/option not available	503*	Service unavailable

* Messages and responses were created since the 'ISUP to SIP Mapping' draft doesn't specify their cause code mapping.

Table H-1: Mapping of ISDN Release Reason to SIP Response (continues on pages 380 to 381)

ISDN Release Reason	Description	SIP Response	Description
65	Bearer capability not implemented	501	Not implemented
66	Channel type not implemented	480*	Temporarily unavailable
69	Requested facility not implemented	503*	Service unavailable
70	Only restricted digital information bearer capability is available	503*	Service unavailable
79	Service or option not implemented	501	Not implemented
81	Invalid call reference value	502*	Bad gateway
82	Identified channel does not exist	502*	Bad gateway
83	Suspended call exists, but this call identity does not	503*	Service unavailable
84	Call identity in use	503*	Service unavailable
85	No call suspended	503*	Service unavailable
86	Call having the requested call identity has been cleared	408*	Request timeout
87	User not member of CUG	503	Service unavailable
88	Incompatible destination	503	Service unavailable
91	Invalid transit network selection	502*	Bad gateway
95	Invalid message	503	Service unavailable
96	Mandatory information element is missing	409*	Conflict
97	Message type non-existent or not implemented	480*	Temporarily not available
98	Message not compatible with call state or message type non-existent or not implemented	409*	Conflict
99	Information element non-existent or not implemented	480*	Not found
100	Invalid information elements contents	501*	Not implemented
101	Message not compatible with call state	503*	Service unavailable
102	Recovery of timer expiry	408	Request timeout
111	Protocol error	500	Server internal error
127	Interworking unspecified	500	Server internal error

H.3 Fixed Mapping of SIP Response to ISDN Release Reason

Table H-2 below describes the mapping of SIP response to ISDN release reason.

Table H-2: Mapping of SIP Response to ISDN Release Reason

SIP Response	Description	ISDN Release Reason	Description
400*	Bad request	31	Normal, unspecified
401	Unauthorized	21	Call rejected
402	Payment required	21	Call rejected
403	Forbidden	21	Call rejected
404	Not found	1	Unallocated number
405	Method not allowed	63	Service/option unavailable
406	Not acceptable	79	Service/option not implemented
407	Proxy authentication required	21	Call rejected
408	Request timeout	102	Recovery on timer expiry
409	Conflict	41	Temporary failure
410	Gone	22	Number changed w/o diagnostic
411	Length required	127	Interworking
413	Request entity too long	127	Interworking
414	Request URI too long	127	Interworking
415	Unsupported media type	79	Service/option not implemented
420	Bad extension	127	Interworking
480	Temporarily unavailable	18	No user responding
481*	Call leg/transaction doesn't exist	127	Interworking
482*	Loop detected	127	Interworking
483	Too many hops	25	Exchange – routing error
484	Address incomplete	28	Invalid number format
485	Ambiguous	1	Unallocated number
486	Busy here	17	User busy
488	Not acceptable here	31	Normal, unspecified
500	Server internal error	41	Temporary failure
501	Not implemented	38	Network out of order
502	Bad gateway	38	Network out of order
503	Service unavailable	41	Temporary failure
504	Server timeout	102	Recovery on timer expiry
505*	Version not supported	127	Interworking
600	Busy everywhere	17	User busy
603	Decline	21	Call rejected
604	Does not exist anywhere	1	Unallocated number
606*	Not acceptable	38	Network out of order

* Messages and responses were created since the 'ISUP to SIP Mapping' draft doesn't specify their cause code mapping.

I SNMP Traps

This section provides information on proprietary SNMP traps currently supported by the gateway. There is a separation between traps that are alarms and traps that are not (logs). Currently all have the same structure made up of the same 11 varbinds (Variable Binding) (1.3.6.1.4.1.5003.9.10.1.21.1).

The source varbind is composed of a string that details the component from which the trap is being sent (forwarded by the hierarchy in which it resides). For example, an alarm from an SS7 link has the following string in its source varbind:

`acBoard#1/SS7#0/SS7Link#6`

In this example, the SS7 link number is specified as 6 and is part of the only SS7 module in the device that is placed in slot number 1 (in a chassis) and is the module to which this trap relates. For devices where there are no chassis options the slot number of the gateway is always 1.

I.1 Alarm Traps

The following tables provide information on alarms that are raised as a result of a generated SNMP trap. The component name (described in each of the following headings) refers to the string that is provided in the 'acBoardTrapGlobalsSource' trap varbind. To clear a generated alarm the same notification type is sent but with the severity set to 'cleared'.

I.1.1 Component: Board#<n>

The source varbind text for all the alarms under this component is Board#<n> where n is the slot number when the board resides in a chassis (for TP-260 and TP-1610, <n> = 1).

Table I-1: acBoardFatalError Alarm Trap

Alarm:	acBoardFatalError
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.1
Default Severity	Critical
Event Type:	equipmentAlarm
Probable Cause:	underlyingResourceUnavailable (56)
Alarm Text:	Board Fatal Error: <text>
Status Changes:	
Condition:	Any fatal error
Alarm status:	Critical
<text> value:	A run-time specific string describing the fatal error
Condition:	After fatal error
Alarm status:	Status stays critical until reboot. A clear trap is not sent.
Corrective Action:	Capture the alarm information and the Syslog clause, if active. Contact your first-level support group. The support group will likely want to collect additional data from the device and perform a reset.

Table I-2: acBoardConfigurationError Alarm Trap

Alarm:	acBoardConfigurationError
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.2
Default Severity	Critical
Event Type:	equipmentAlarm
Probable Cause:	underlyingResourceUnavailable (56)
Alarm Text:	Board Config Error: <text>
Status Changes:	
Condition:	A configuration error was detected
Alarm status:	critical
<text> value:	A run-time specific string describing the configuration error.
Condition:	After configuration error
Alarm status:	Status stays critical until reboot. A clear trap is not sent.
Corrective Action:	Inspect the run-time specific string to determine the nature of the configuration error. Fix the configuration error using the appropriate tool: Web interface, EMS, or *ini* file. Save the configuration and if necessary reset the device.

Table I-3: acBoardTemperatureAlarm Alarm Trap

Alarm:	acBoardTemperatureAlarm (doesn't apply to the TP-260 board)
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.3
Default Severity	Critical
Event Type:	equipmentAlarm
Probable Cause:	temperatureUnacceptable (50)
Alarm Text:	Board temperature too high
Status Changes:	
Condition:	Temperature is above 60 degrees C (140 degrees F)
Alarm status:	Critical
Condition:	After raise, temperature falls below 55 degrees C (131 degrees F)
Alarm status:	Cleared
Corrective Action:	Inspect the system. Determine if all fans in the system are properly operating.

Table I-4: acBoardEvResettingBoard Alarm Trap

Alarm:	acBoardEvResettingBoard
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.5
Default Severity	Critical
Event Type:	equipmentAlarm
Probable Cause:	outOfService (71)
Alarm Text:	User resetting board
Status Changes:	
Condition:	When a soft reset is triggered via the Web interface or SNMP.
Alarm status:	Critical
Condition:	After raise
Alarm status:	Status stays critical until reboot. A clear trap is not sent.
Corrective Action:	A network administrator has taken action to reset the device. No corrective action is required.

Table I-5: acFeatureKeyError Alarm Trap

Alarm:	acFeatureKeyError
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.6
Default Severity	Critical
Event Type:	processingErrorAlarm
Probable Cause:	configurationOrCustomizationError (7)
Alarm Text:	Feature key error
Status Changes:	
Note:	Support of this alarm is pending

Table I-6: acBoardCallResourcesAlarm Alarm Trap

Alarm:	acBoardCallResourcesAlarm
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.8
Default Severity	Major
Event Type:	processingErrorAlarm
Probable Cause:	softwareError (46)
Alarm Text:	Call resources alarm
Status Changes:	
Condition:	Number of free channels exceeds the predefined RAI *high* threshold.
Alarm Status:	Major
Note:	To enable this alarm the RAI mechanism must be activated (EnableRAI = 1).
Condition:	Number of free channels falls below the predefined RAI *low* threshold.
Alarm Status:	Cleared

Table I-7: acBoardControllerFailureAlarm Alarm Trap

Alarm:	acBoardControllerFailureAlarm
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.9
Default Severity	Minor
Event Type:	processingErrorAlarm
Probable Cause:	softwareError (46)
Alarm Text:	Controller failure alarm
Status Changes:	
Condition:	Proxy has not been found
Alarm Status:	Major
Additional Info:	Proxy not found. Use internal routing or Proxy lost. looking for another Proxy
Condition:	Proxy is found. The clear message includes the IP address of this Proxy.
Alarm Status:	Cleared

Table I-8: acBoardOverloadAlarm Alarm Trap

Alarm:	acBoardOverloadAlarm
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.11
Default Severity	Major
Event Type:	processingErrorAlarm
Probable Cause:	softwareError (46)
Alarm Text:	Board overload alarm
Status Changes:	
Condition:	An overload condition exists in one or more of the system components.
Alarm Status:	Major
Condition:	The overload condition passed
Alarm Status:	Cleared

I.1.2 Component: AlarmManager#0

The source varbind text for all the alarms under this component is *Board#*<n>*/AlarmManager#0* where n is the slot number.

Table I-9: acActiveAlarmTableOverflow Alarm Trap

Alarm:	acActiveAlarmTableOverflow
OID:	1.3.6.1.4.15003.9.10.1.21.2.0.12
Default Severity	Major
Event Type:	processingErrorAlarm
Probable Cause:	resourceAtOrNearingCapacity (43)
Alarm Text:	Active alarm table overflow
Status Changes:	
Condition:	Too many alarms to fit in the active alarm table
Alarm status:	major
Condition:	After raise
Alarm status:	Status stays major until reboot. A clear trap is not sent.
Note:	The status stays major until reboot as it denotes a possible loss of information until the next reboot. If an alarm is raised when the table is full, it is possible that the alarm is active, but does not appear in the active alarm table.
Corrective Action:	Some alarm information may have been lost, but the ability of the device to perform its basic operations has not been impacted. A reboot is the only way to completely clear a problem with the active alarm table. Contact your first-level group.

I.1.3 Component: EthernetLink#0

The source varbind text for all the alarms under this component is *Board#*<n>*/EthernetLink#0* where n is the slot number.

This trap relates to the Ethernet Link Module (the #0 numbering doesn't apply to the physical Ethernet link). This trap doesn't apply to TP-260 boards.

Table I-10: acBoardEthernetLinkAlarm Alarm Trap

Alarm:	acBoardEthernetLinkAlarm
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.10
Default Severity	Critical
Event Type:	equipmentAlarm
Probable Cause:	underlyingResourceUnavailable (56)
Alarm Text:	Ethernet link alarm: <text>
Status Changes:	
Condition:	Fault on single interface
Alarm status:	major
<text> value:	Redundant link is down
Condition:	Fault on both interfaces
Alarm status:	critical
<text> value:	No Ethernet link
Condition:	Both interfaces are operational
Alarm status:	cleared
Corrective Action:	Ensure that both Ethernet cables are plugged into the back of the system. Inspect the system's Ethernet link lights to determine which interface is failing. Reconnect the cable or fix the network problem

I.1.4 Component: SS7#0

The source varbind text for all alarms under this component is Board#<n>/SS7#0/SS7Link#<m> where n is the slot number and m is the link number.

Table I-11: acSS7LinkStateChangeAlarm Trap

Alarm:	acSS7LinkStateChangeAlarm
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.19
Default Severity	Major
Event Type:	communicationsAlarm
Probable Cause:	Other
Alarm Text:	*** SS7 *** Link %i is %s $s
Status Changes:	
Condition:	Operational state of the SS7 link becomes 'BUSY'.
Alarm status:	Major
<text> value:	%i - <Link number> %s - <state name>: { "OFFLINE", "BUSY", "INSERVICE"} %s – If link has MTP3 layer, then this string equals: (SP %i linkset %i slc %i) Where: %i - <SP number> %i - <Link-Set number> %i - <SLC number> Otherwise there is NO additional text.
Additional Info1 varbind	BUSY
Condition:	Operational state of the link becomes 'IN-SERVICE' or 'OFFLINE'.
Alarm status:	Cleared
Corrective Action:	For full details refer to the SS7 MTP2 and MTP3 relevant standards.

Table I-12: acSS7LinkCongestionStateChangeAlarm Trap

Alarm:	acSS7LinkCongestionStateChangeAlarm
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.22
Default Severity	Major
Event Type:	communicationsAlarm
Probable Cause:	Other
Alarm Text:	*** SS7 *** Link %i is %s %s
Status Changes:	
Condition:	SS7 link becomes congested (local or remote).
Alarm status:	Major
<text> value:	%i - <Link number> %s – If link has MTP3 layer, then this string equals: (SP %i linkset %i slc %i) Where: %i - <SP number> %i - <Link-Set number> %i - <SLC number> Otherwise there is NO additional text. %s - <congestion state>: { "UNCONGESTED", "CONGESTED" }
Additional Info1 varbind	CONGESTED
Condition:	Link becomes un-congested (local AND remote).
Alarm status:	Cleared
Corrective Action:	Reduce SS7 traffic on that link.
Note:	This alarm is raised for any change in the remote or local congestion status.

I.1.5 Log Traps (Notifications)

This section details traps that are not alarms. These traps are sent with the severity varbind value of 'indeterminate'. These traps don't 'clear', they don't appear in the alarm history or active tables. One log trap that does send clear is acPerformanceMonitoringThresholdCrossing.

Table I-13: acKeepAlive Log Trap

Trap:	acKeepAlive
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.16
Default Severity	Indeterminate
Event Type:	other (0)
Probable Cause:	other (0)
Trap Text:	Keep alive trap
Status Changes:	
Condition:	The STUN client in is enabled and identified a NAT device or doesn't locate the STUN server. The *ini* file contains the following line: 'SendKeepAliveTrap=1'
Trap status:	Trap is sent
Note:	Keep-alive is sent every 9/10 of the time defined in the parameter NatBindingDefaultTimeout.

Table I-14: acPerformanceMonitoringThresholdCrossing Log Trap

Trap:	acPerformanceMonitoringThresholdCrossing
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.27
Default Severity	Indeterminate
Event Type:	other (0)
Probable Cause:	other (0)
Trap Text:	"Performance: Threshold trap was set", with source = name of performance counter which caused the trap
Status Changes:	
Condition:	A performance counter has crossed the high threshold
Trap status:	Indeterminate
Condition:	A performance counter has crossed the low threshold
Trap status:	cleared

Table I-15: acHTTPDownloadResult Log Trap

Trap:	acHTTPDownloadResult
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.28
Default Severity	Indeterminate
Event Type:	processingErrorAlarm (3) for failures and other (0) for success.
Probable Cause:	other (0)
Status Changes:	
Condition:	Successful HTTP download.
Trap text:	HTTP Download successful
Condition:	Failed download.
Trap text:	HTTP download failed, a network error occurred.
Note:	There are other possible textual messages describing NFS failures or success, FTP failure or success.

I.1.6 Other Traps

The following are provided as SNMP traps and are not alarms.

Table I-16: coldStart Trap

Trap Name:	coldStart
OID:	1.3.6.1.6.3.1.1.5.1
MIB	SNMPv2-MIB
Note:	This is a trap from the standard SNMP MIB.

Table I-17: authenticationFailure Trap

Trap Name:	authenticationFailure
OID:	1.3.6.1.6.3.1.1.5.5
MIB	SNMPv2-MIB

Table I-18: acBoardEvBoardStarted Trap

Trap Name:	acBoardEvBoardStarted
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.4
MIB	AcBoard
Severity	cleared
Event Type:	equipmentAlarm
Probable Cause:	Other(0)
Alarm Text:	Initialization Ended
Note:	This is the AudioCodes Enterprise application cold start trap.

Table I-19: AcDChannelStatus Trap

Trap Name:	acDChannelStatus
OID:	1.3.6.1.4.1.5003.9.10.1.21.2.0.37
MIB	AcBoard
Severity	Minor
Event Type:	communicationsAlarm
Probable Cause:	communicationsProtocolError
Alarm Text:	D-Channel Trap.
Source:	Trunk <m> where m is the trunk number (starts from 0).
Status Changes:	
Condition:	D-Channel un-established.
Trap Status:	Trap is sent with the severity of Minor.
Condition:	D-Channel established.
Trap Status:	Trap is sent with the severity of Cleared.

I.1.7 Trap Varbinds

Each trap described above provides the following fields (known as 'varbinds'). Refer to the AcBoard MIB for additional details on these varbinds.

- acBoardTrapGlobalsName
- acBoardTrapGlobalsTextualDescription
- acBoardTrapGlobalsSource
- acBoardTrapGlobalsSeverity
- acBoardTrapGlobalsUniqID
- acBoardTrapGlobalsType
- acBoardTrapGlobalsDateAndTime
- acBoardTrapGlobalsProbableCause
- acBoardTrapGlobalsAdditionalInfo1
- acBoardTrapGlobalsAdditionalInfo2
- acBoardTrapGlobalsAdditionalInfo3

Note that 'acBoardTrapGlobalsName' is actually a number. The value of this varbind is 'X' minus 1, where 'X' is the last number in the trap's OID. For example, the 'name' of 'acBoardEthernetLinkAlarm' is '9'. The OID for 'acBoardEthernetLinkAlarm' is 1.3.6.1.4.1.5003. 9.10.1.21.2.0.10.

J Installation and Configuration of Apache HTTP Server

This appendix describes the installation and configuration of Apache's HTTP server with Perl script environment (required for recording).

J.1 Windows 2000/XP Operation Systems

Note: For detailed installation information, refer to http://perl.apache.org/docs/2.0/os/win32/config.html.

➢ To configure the Apache HTTP server and mod_perl version 2.0 software, take these 9 steps:

Additional required software: an uploading script (put.cgi), supplied with the software package.

1. Download the third party Perl-5.8-win32-bin.exe, installation file, from the following link: www.apache.org/dist/perl/win32-bin/Perl-5.8-win32-bin.exe. The installation file includes: Apache 2.0.46, Perl 5.8.0 and mod_perl-1.99 (the content of the file and the software version are subject to modification and changes in the future).

   For full installation instructions refer to www.apache.org/dist/perl/win32-bin/Perl-5.8-win32-bin.readme.
2. To start the installation wizard run the Perl-5.8-win32-bin.exe file.
3. During the installing, you are prompt to determine the Destination Folder under which the package is installed, it is advised to provide a non-spaced path (such as: c:\\directory_name_without_spaces).
4. In the following screen (configuration): uncheck the "Build html docs" and "Configure CPAN pm" checkboxes, if they are present. If you are prompted to bring "nmake" answer no.
5. After the installation is complete, add the "/path/perl/bin" and "/path/apache2/bin" (path stands for the path that was previously specified in the "Destination Folder") directories to the system known path. Open the Control Panel→System→Advanced→Environment Variables, inside the System Variables dialog box choose "Path" and click the Edit button; in the opened Variable Value checkbox append both of the paths to the existing list. Restart window in order to activate the new paths.
6. Open the Apache2/conf/httpd.conf file for editing and set the parameter MaxKeepAliveRequests to 0 (enables an unlimited number of requests during a persistent connection – required for multiple consecutive HTTP PUT requests for uploading the file).

7. Open the Apache2/conf/perl.conf file for editing and add the line "Script PUT /perl/put.cgi" after the last line in the following section (note that if the following section is omitted or different in the file, insert it into the file or change it there accordingly):

```
Alias /perl/ "C:/Apache2/perl/
<Location /perl>
SetHandler perl-script
PerlResponseHandler ModPerl::Registry
Options +ExecCGI
PerlOptions +ParseHeaders
</Location>
```

8. Locate the file put.cgi on the supplied software package and copy it into the Apach2\perl\ directory. Change the first line in this file from c:/perl/bin/perl to your perl executable file (this step is required only if mod_perl is not included in your Apache installation).
9. In the apach2\bin directory, from a DOS prompt, type the following commands:
   a. Apache.exe -n Apache2 -k install
   b. Apache.exe -n Apache2 -k start

The installation and configuration are finished. You are now ready to start using the HTTP server.

J.2 Linux Operation Systems

> **Note:** It is assumed that the installing of Linux already includes: Apache server (for example, Apache 1.3.23), perl and mod_perl (for example mod_perl 1.26).

➢ To configure Apache HTTP server, take these 4 steps:

Additional required software: an uploading script (put.cgi), supplied with the software package.

1. Inside the Apache directory, create the directory /perl (for example /var/www/perl). Locate the file put.cgi on the supplied software package and copy it to that directory.
2. In the put.cgi script, change the first line from c:/perl/bin/perl to your perl executable file (this step is required only if mod_perl is not included in your Apache installation).
3. Enable access to the following directories and files by typing:
   - >chmod 777 perl
   - >chmod 755 put.cgi
   - >chmod 777 html (the name of the server's shared files directory)

4. Configure the Apache sever:
   a. Open etc/httpd/conf/httpd.conf (or a similar file) for editing
   b. Set the KeepAlive parameter to true
   c. Set the MaxKeepAliveRequests parameter to 0 (enables an unlimited number of requests during a persistent connection – required for multiple consecutive HTTP POST requests for uploading the file).
   d. Set MaxClients to 250
   e. Change the mod_perl module lines to:

```
<IfModule mod perl.c>
    Alias /perl/ /var/www/perl/
    <Directory /var/www/perl>
        SetHandler perl-script
        PerlHandler Apache::Registry
        Options +ExecCGI
        PerlSendHeader On
    </Directory>
</IfModule>

Script PUT /perl/put.cgi
```

Reader's Notes

K Regulatory Information

K.1 Mediant 2000

Declaration of Conformity

Application of Council Directives:	73/23/EEC (including amendments) 89/336/EEC (including amendments) 1999/5/EC Annex-II of the Directive
Standards to which Conformity is Declared:	EN55022: 1998 + A1: 2000 + A2: 2003 EN55024:1998 + A1: 2001 + A2: 2003 EN61000-3-2: 2000 + A2: 2005 (AC only) EN61000-3-3: 1995 + A1: 2001 (AC only) EN60950-1: 2001
Manufacturer's Name:	AudioCodes Ltd.
Manufacturer's Address:	1 Hayarden Street, Airport City, Lod 70151, Israel.
Type of Equipment:	Digital VoIP System
Model Numbers:	**Mediant 2000**, **Stretto 2000, IPmedia 2000**

I, the undersigned, hereby declare that the equipment specified above conforms to the above Directives and Standards.

	17th April, 2006	Airport City, Lod, Israel
Signature	*Date (Day/Month/Year)*	*Location*

I. Zusmanovich, Compliance Engineering Manager

Czech	[AudioCodes Ltd] tímto prohlašuje, že tento [2000 Series] je ve shodě se základními požadavky a dalšími příslušnými ustanoveními směrnice 1999/5/ES."
Danish	Undertegnede [AudioCodes Ltd] erklærer herved, at følgende udstyr [2000 Series] overholder de væsentlige krav og øvrige relevante krav i direktiv 1999/5/EF
Dutch	Hierbij verklaart [AudioCodes Ltd] dat het toestel [2000 Series] in overeenstemming is met de essentiële eisen en de andere relevante bepalingen van richtlijn 1999/5/EG
English	Hereby, [AudioCodes Ltd], declares that this [2000 Series] is in compliance with the essential requirements and other relevant provisions of Directive 1999/5/EC.
Estonian	Käesolevaga kinnitab [AudioCodes Ltd] seadme [2000 Series] vastavust direktiivi 1999/5/EÜ põhinõuetele ja nimetatud direktiivist tulenevatele teistele asjakohastele sätetele.
Finnish	[AudioCodes Ltd] vakuuttaa täten että [2000 Series] tyyppinen laite on direktiivin 1999/5/EY oleellisten vaatimusten ja sitä koskevien direktiivin muiden ehtojen mukainen.
French	Par la présente [AudioCodes Ltd] déclare que l'appareil [2000 Series] est conforme aux exigences essentielles et aux autres dispositions pertinentes de la directive 1999/5/CE
German	Hiermit erklärt [AudioCodes Ltd], dass sich dieser/diese/dieses [2000 Series] in Übereinstimmung mit den grundlegenden Anforderungen und den anderen relevanten Vorschriften der Richtlinie 1999/5/EG befindet". (BMWi)
Greek	ΜΕ ΤΗΝ ΠΑΡΟΥΣΑ [AudioCodes Ltd] ΔΗΛΩΝΕΙ ΟΤΙ [2000 Series] ΣΥΜΜΟΡΦΩΝΕΤΑΙ ΠΡΟΣ ΤΙΣ ΟΥΣΙΩΔΕΙΣ ΑΠΑΙΤΗΣΕΙΣ ΚΑΙ ΤΙΣ ΛΟΙΠΕΣ ΣΧΕΤΙΚΕΣ ΔΙΑΤΑΞΕΙΣ ΤΗΣ ΟΔΗΓΙΑΣ 1999/5/ΕΚ
Hungarian	Alulírott, [AudioCodes Ltd] nyilatkozom, hogy a [2000 Series] megfelel a vonatkozó alapvetõ követelményeknek és az 1999/5/EC irányelv egyéb elõírásainak
Icelandic	æki þetta er í samræmi við tilskipun Evrópusambandsins 1999/5
Italian	Con la presente [AudioCodes Ltd] dichiara che questo [2000 Series] è conforme ai requisiti essenziali ed alle altre disposizioni pertinenti stabilite dalla direttiva 1999/5/CE.
Latvian	Ar šo [AudioCodes Ltd] deklarē, ka [2000 Series] atbilst Direktīvas 1999/5/EK būtiskajām prasībām un citiem ar to saistītajiem noteikumiem.
Lithuanian	[AudioCodes Ltd] deklaruoja, kad irenginys [2000 Series] tenkina 1999/5/EB Direktyvos esminius reikalavimus ir kitas sios direktyvos nuostatas
Maltese	Hawnhekk, [AudioCodes Ltd], jiddikjara li dan [2000 Series] jikkonforma mal-ħtiġijiet essenzjali u ma provvedimenti oħrajn relevanti li hemm fid-Dirrettiva 1999/5/EC
Norwegian	Dette produktet er i samhørighet med det Europeiske Direktiv 1999/5
Polish	[AudioCodes Ltd], deklarujemy z pelna odpowiedzialnoscia, ze wyrób [2000 Series] spelnia podstawowe wymagania i odpowiada warunkom zawartym w dyrektywie 1999/5/EC
Portuguese	[AudioCodes Ltd] declara que este [2000 Series] está conforme com os requisitos essenciais e outras disposições da Directiva 1999/5/CE.
Slovak	[AudioCodes Ltd] týmto vyhlasuje, že [2000 Series] spĺňa základné požiadavky a všetky príslušné ustanovenia Smernice 1999/5/ES.
Slovene	Šiuo [AudioCodes Ltd] deklaruoja, kad šis [2000 Series] atitinka esminius reikalavimus ir kitas 1999/5/EB Direktyvos nuostatas.
Spanish	Por medio de la presente [AudioCodes Ltd] declara que el [2000 Series] cumple con los requisitos esenciales y cualesquiera otras disposiciones aplicables o exigibles de la Directiva 1999/5/CE
Swedish	Härmed intygar [AudioCodes Ltd] att denna [2000 Series] står I överensstämmelse med de väsentliga egenskapskrav och övriga relevanta bestämmelser som framgår av direktiv 1999/5/EG.

Safety Notices

1. Installation and service of this gateway must only be performed by authorized, qualified service personnel.
2. The protective earth terminal on the device must be permanently connected to protective earth.
3. The equipment must be connected by service personnel to a socket-outlet with a protective earthing connection.
4. This equipment should be installed in restricted access locations with maximum allowed temperature 40°C (104°F).

Industry Canada Notice

This equipment meets the applicable Industry Canada Terminal Equipment technical specifications. This is confirmed by the registration numbers. The abbreviation, IC, before the registration number signifies that registration was performed based on a declaration of conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

Digital Device Warnings

This equipment complies with Part 68 of the FCC rules and the requirements adopted by ACTA. On the bottom of this equipment is a label that contains a product identifier in the format US: AC1ISNANTP1610. If requested this number must be provided to the telephone company.

The Telephone company may make changes in the facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service. Should you experience trouble with this telephone equipment, contact: *AudioCodes Inc, San Jose, CA USA. Tel: 1 408 441 1175.* **Do not attempt to repair this equipment!**

Facility Interface Code: **04DU9.BN, 04DU9.DN, 04DU9.1KN, 4DU9.ISN**
Service Order Code: **6.0F**
USOC Jack Type: **RJ21X or RJ48C**

It this gateway causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also you will be advised of your right to file complaint with the FCC if you believe it is necessary.

Network Information and Intent of Use

The products are for access to ISDN at 2048 kb/s and for access to G.703 Leased lines at 2048 kb/s.

Network Compatibility

The products support the Telecom networks in EU that comply with TBR4 and TBR13.

Telecommunication Safety

The safety status of each port is declared and detailed in the table below:

Ports	Safety Status
E1 or T1	TNV-1
Ethernet (100 Base-TX)	SELV
DC input port (applicable only when DC powered)	SELV

TNV-1: Telecommunication network voltage circuits whose normal operating voltages do not exceed the limits for SELV under normal operating conditions and on which over voltages from telecommunication networks are possible.
SELV: Safety extra low voltage circuit.

FCC Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

K.2 TP-1610

Declaration of Conformity

Application of Council Directives:	73/23/EEC (including amendments) 89/336/EEC (including amendments) 1999/5/EC Annex-II of the Directive
Standards to which Conformity is Declared:	EN55022: 1998 + A1: 2000 + A2: 2003 EN55024:1998 + A1: 2001 + A2: 2003 EN60950-1: 2001
Manufacturer's Name:	AudioCodes Ltd.
Manufacturer's Address:	1 Hayarden Street, Airport City, Lod 70151, Israel.
Type of Equipment:	Digital VoIP System
Model Numbers:	**TP-1610, IPM-1610, SB-1610**

I, the undersigned, hereby declare that the equipment specified above conforms to the above Directives and Standards.

	27th June, 2006	Airport City, Lod, Israel
Signature	*Date (Day/Month/Year)*	*Location*

I. Zusmanovich, Compliance Engineering Manager

Czech	[AudioCodes Ltd] tímto prohlašuje, že tento [1610 series] je ve shodě se základními požadavky a dalšími příslušnými ustanoveními směrnice 1999/5/ES."
Danish	Undertegnede [AudioCodes Ltd] erklærer herved, at følgende udstyr [1610 Series] overholder de væsentlige krav og øvrige relevante krav i direktiv 1999/5/EF
Dutch	Hierbij verklaart [AudioCodes Ltd] dat het toestel [1610 Series] in overeenstemming is met de essentiële eisen en de andere relevante bepalingen van richtlijn 1999/5/EG
English	Hereby, [AudioCodes Ltd], declares that this [1610 Series] is in compliance with the essential requirements and other relevant provisions of Directive 1999/5/EC.
Estonian	Käesolevaga kinnitab [AudioCodes Ltd] seadme [1610 Series] vastavust direktiivi 1999/5/EÜ põhinõuetele ja nimetatud direktiivist tulenevatele teistele asjakohastele sätetele.
Finnish	[AudioCodes Ltd] vakuuttaa täten että [1610 Series] tyyppinen laite on direktiivin 1999/5/EY oleellisten vaatimusten ja sitä koskevien direktiivin muiden ehtojen mukainen.
French	Par la présente [AudioCodes Ltd] déclare que l'appareil [1610 Series] est conforme aux exigences essentielles et aux autres dispositions pertinentes de la directive 1999/5/CE
German	Hiermit erklärt [AudioCodes Ltd], dass sich dieser/diese/dieses [1610 Series] in Übereinstimmung mit den grundlegenden Anforderungen und den anderen relevanten Vorschriften der Richtlinie 1999/5/EG befindet". (BMWi)
Greek	ΜΕ ΤΗΝ ΠΑΡΟΥΣΑ [AudioCodes Ltd] ΔΗΛΩΝΕΙ ΟΤΙ [1610 Series] ΣΥΜΜΟΡΦΩΝΕΤΑΙ ΠΡΟΣ ΤΙΣ ΟΥΣΙΩΔΕΙΣ ΑΠΑΙΤΗΣΕΙΣ ΚΑΙ ΤΙΣ ΛΟΙΠΕΣ ΣΧΕΤΙΚΕΣ ΔΙΑΤΑΞΕΙΣ ΤΗΣ ΟΔΗΓΙΑΣ 1999/5/ΕΚ
Hungarian	Alulírott, [AudioCodes Ltd] nyilatkozom, hogy a [1610 Series] megfelel a vonatkozó alapvetõ követelményeknek és az 1999/5/EC irányelv egyéb elõírásainak
Icelandic	æki þetta er í samræmi við tilskipun Evrópusambandsins 1999/5
Italian	Con la presente [AudioCodes Ltd] dichiara che questo (1610 Series) è conforme ai requisiti essenziali ed alle altre disposizioni pertinenti stabilite dalla direttiva 1999/5/CE.
Latvian	Ar šo [AudioCodes Ltd] deklarē, ka [1610 Series] atbilst Direktīvas 1999/5/EK būtiskajām prasībām un citiem ar to saistītajiem noteikumiem.
Lithuanian	[AudioCodes Ltd] deklaruoja, kad irenginys [1610 Series] tenkina 1999/5/EB Direktyvos esminius reikalavimus ir kitas sios direktyvos nuostatas
Maltese	Hawnhekk, [AudioCodes Ltd], jiddikjara li dan [1610 Series] jikkonforma mal-ħtiġijiet essenzjali u ma provvedimenti oħrajn relevanti li hemm fid-Dirrettiva 1999/5/EC
Norwegian	Dette produktet er i samhørighet med det Europeiske Direktiv 1999/5
Polish	[AudioCodes Ltd], deklarujemy z pelna odpowiedzialnoscia, ze wyrób [1610 Series] spelnia podstawowe wymagania i odpowiada warunkom zawartym w dyrektywie 1999/5/EC
Portuguese	[AudioCodes Ltd] declara que este [1610 Series] está conforme com os requisitos essenciais e outras disposições da Directiva 1999/5/CE.
Slovak	[AudioCodes Ltd] týmto vyhlasuje, že [1610 Series] spĺňa základné požiadavky a všetky príslušné ustanovenia Smernice 1999/5/ES.
Slovene	Šiuo [AudioCodes Ltd] deklaruoja, kad šis [1610 Series] atitinka esminius reikalavimus ir kitas 1999/5/EB Direktyvos nuostatas.
Spanish	Por medio de la presente [AudioCodes Ltd] declara que el (1610 Series) cumple con los requisitos esenciales y cualesquiera otras disposiciones aplicables o exigibles de la Directiva 1999/5/CE
Swedish	Härmed intygar [AudioCodes Ltd] att denna [1610 Series] står I överensstämmelse med de väsentliga egenskapskrav och övriga relevanta bestämmelser som framgår av direktiv 1999/5/EG.

Safety Notice

Installation and service of this gateway must only be performed by authorized, qualified service personnel.

Industry Canada Notice

This equipment meets the applicable Industry Canada Terminal Equipment technical specifications. This is confirmed by the registration numbers. The abbreviation, IC, before the registration number signifies that registration was performed based on a declaration of conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

Digital Device Warnings

This equipment complies with Part 68 of the FCC rules and the requirements adopted by ACTA. On the interface card module of this equipment is a label that contains a product identifier in the format US: AC1ISNANTP1610. If requested this number must be provided to the telephone company.

The Telephone Company may make changes in the facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service. Should you experience trouble with this telephone equipment, contact: *AudioCodes Inc, San Jose, CA USA. Tel: 1 408 441 1175.* **Do not attempt to repair this equipment!**

Facility Interface Code: **04DU9.BN, 04DU9.DN, 04DU9.1KN, 4DU9.ISN**
Service Order Code: **6.0F**
USOC Jack Type: **RJ21X or RJ48C**

It this gateway causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also you will be advised of your right to file complaint with the FCC if you believe it is necessary.

Network Information and Intent of Use

The products are for access to ISDN at 2048 kb/s and for access to G.703 Leased lines at 2048 kb/s.

Network Compatibility

The products support the Telecom networks in EU that comply with TBR4 and TBR13.

Telecommunication Safety

The safety status of each port is declared and detailed in the table below:

Ports	Safety Status
E1 or T1	TNV-1
Ethernet (100 Base-TX)	SELV

TNV-1: Telecommunication network voltage circuits whose normal operating voltages do not exceed the limits for SELV under normal operating conditions and on which over voltages from telecommunication networks are possible.
SELV: Safety extra low voltage circuit.

FCC Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

K.3 TP-260

Declaration of Conformity

Application of Council Directives:	73/23/EEC (including amendments) 89/336/EEC (including amendments) 1999/5/EC Annex-II of the Directive
Standards to which Conformity is Declared:	EN55022: 1998 + A1: 2000 + A2: 2003 EN55024:1998 + A1: 2001 + A2: 2003 EN60950-1: 2001
Manufacturer's Name:	AudioCodes Ltd.
Manufacturer's Address:	1 Hayarden Street, Airport City, Lod 70151, Israel.
Type of Equipment:	Digital VoIP Board
Model Numbers:	**TP-260/UNI, IPM-260/UNI**

I, the undersigned, hereby declare that the equipment specified above conforms to the above Directives and Standards.

	27th June, 2006	Airport City, Lod, Israel
Signature	*Date (Day/Month/Year)*	*Location*

I. Zusmanovich, Compliance Engineering Manager

Czech	[AudioCodes Ltd] tímto prohlašuje, že tento [260 series] je ve shodě se základními požadavky a dalšími příslušnými ustanoveními směrnice 1999/5/ES."
Danish	Undertegnede [AudioCodes Ltd] erklærer herved, at følgende udstyr [260 Series] overholder de væsentlige krav og øvrige relevante krav i direktiv 1999/5/EF
Dutch	Hierbij verklaart [AudioCodes Ltd] dat het toestel [260 Series] in overeenstemming is met de essentiële eisen en de andere relevante bepalingen van richtlijn 1999/5/EG
English	Hereby, [AudioCodes Ltd], declares that this [260 Series] is in compliance with the essential requirements and other relevant provisions of Directive 1999/5/EC.
Estonian	Käesolevaga kinnitab [AudioCodes Ltd] seadme [260 Series] vastavust direktiivi 1999/5/EÜ põhinõuetele ja nimetatud direktiivist tulenevatele teistele asjakohastele sätetele.
Finnish	[AudioCodes Ltd] vakuuttaa täten että [260 Series] tyyppinen laite on direktiivin 1999/5/EY oleellisten vaatimusten ja sitä koskevien direktiivin muiden ehtojen mukainen.
French	Par la présente [AudioCodes Ltd] déclare que l'appareil [260 Series] est conforme aux exigences essentielles et aux autres dispositions pertinentes de la directive 1999/5/CE
German	Hiermit erklärt [AudioCodes Ltd], dass sich dieser/diese/dieses [260 Series] in Übereinstimmung mit den grundlegenden Anforderungen und den anderen relevanten Vorschriften der Richtlinie 1999/5/EG befindet". (BMWi)
Greek	ΜΕ ΤΗΝ ΠΑΡΟΥΣΑ [AudioCodes Ltd] ΔΗΛΩΝΕΙ ΟΤΙ [260 Series] ΣΥΜΜΟΡΦΩΝΕΤΑΙ ΠΡΟΣ ΤΙΣ ΟΥΣΙΩΔΕΙΣ ΑΠΑΙΤΗΣΕΙΣ ΚΑΙ ΤΙΣ ΛΟΙΠΕΣ ΣΧΕΤΙΚΕΣ ΔΙΑΤΑΞΕΙΣ ΤΗΣ ΟΔΗΓΙΑΣ 1999/5/ΕΚ
Hungarian	Alulírott, [AudioCodes Ltd] nyilatkozom, hogy a [260 Series] megfelel a vonatkozó alapvetõ követelményeknek és az 1999/5/EC irányelv egyéb elõírásainak
Icelandic	æki þetta er í samræmi við tilskipun Evrópusambandsins 1999/5
Italian	Con la presente [AudioCodes Ltd] dichiara che questo (260 Series) è conforme ai requisiti essenziali ed alle altre disposizioni pertinenti stabilite dalla direttiva 1999/5/CE.
Latvian	Ar šo [AudioCodes Ltd] deklarē, ka [260 Series] atbilst Direktīvas 1999/5/EK būtiskajām prasībām un citiem ar to saistītajiem noteikumiem.
Lithuanian	[AudioCodes Ltd] deklaruoja, kad irenginys [260 Series] tenkina 1999/5/EB Direktyvos esminius reikalavimus ir kitas sios direktyvos nuostatas
Maltese	Hawnhekk, [AudioCodes Ltd], jiddikjara li dan [260 Series] jikkonforma mal-ħtiġijiet essenzjali u ma provvedimenti oħrajn relevanti li hemm fid-Dirrettiva 1999/5/EC
Norwegian	Dette produktet er i samhørighet med det Europeiske Direktiv 1999/5
Polish	[AudioCodes Ltd], deklarujemy z pelna odpowiedzialnoscia, ze wyrób [260 Series] spelnia podstawowe wymagania i odpowiada warunkom zawartym w dyrektywie 1999/5/EC
Portuguese	[AudioCodes Ltd] declara que este [260 Series] está conforme com os requisitos essenciais e outras disposições da Directiva 1999/5/CE.
Slovak	[AudioCodes Ltd] týmto vyhlasuje, že [260 Series] spĺňa základné požiadavky a všetky príslušné ustanovenia Smernice 1999/5/ES.
Slovene	Šiuo [AudioCodes Ltd] deklaruoja, kad šis [260 Series] atitinka esminius reikalavimus ir kitas 1999/5/EB Direktyvos nuostatas.
Spanish	Por medio de la presente [AudioCodes Ltd] declara que el (260 Series) cumple con los requisitos esenciales y cualesquiera otras disposiciones aplicables o exigibles de la Directiva 1999/5/CE
Swedish	Härmed intygar [AudioCodes Ltd] att denna [260 Series] står I överensstämmelse med de väsentliga egenskapskrav och övriga relevanta bestämmelser som framgår av direktiv 1999/5/EG.

Safety Notice

Installation and service of this card must only be performed by authorized, qualified service personnel.

Industry Canada Notice

This equipment meets the applicable Industry Canada Terminal Equipment technical specifications. This is confirmed by the registration numbers. The abbreviation, IC, before the registration number signifies that registration was performed based on a declaration of conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

FCC Digital Device Warnings

This equipment complies with Part 68 of the FCC rules and the requirements adopted by ACTA. On the interface card module of this equipment is a label that contains a product identifier in the format US: AC1ISNANIPM260UNI. If requested this number must be provided to the telephone company.

The Telephone Company may make changes in the facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service. Should you experience trouble with this telephone equipment, contact: *AudioCodes Inc, San Jose, CA USA. Tel: 1 408 441 1175*. **Do not attempt to repair this equipment!**

Facility Interface Code: **04DU9.BN, 04DU9.DN, 04DU9.1KN, 4DU9.ISN**
Service Order Code: **6.0F**
USOC Jack Type: **RJ48C**

It this gateway causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also you will be advised of your right to file complaint with the FCC if you believe it is necessary.

Network Information and Intent of Use

The products are for access to ISDN at 2048 kb/s and for access to G.703 Leased lines at 2048 kb/s.

Network Compatibility

The products support the Telecom networks in EU that comply with TBR4 and TBR13.

Telecommunication Safety

The safety status of each port is declared and detailed in the table below:

Ports	Safety Status
E1 or T1	TNV-1
Ethernet (100 Base-TX)	SELV

TNV-1: Telecommunication network voltage circuits whose normal operating voltages do not exceed the limits for SELV
SELV: Safety extra low voltage circuit.

FCC Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

SIP

Mediant 2000
TP-1610 & TP-260/UNI Boards

User's Manual Version 5.0

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AudioCodes

www.audiocodes.com